Mediant 800 MSBG E-SBC, Mediant 1000 MSBG E-SBC and Mediant 3000 E-SBC Media Gateway

Configuration Note

Connecting AT&T’s IP Flexible Reach - MIS/PNT/AVPN SIP Trunking Service to Microsoft® Lync Server 2010 Using AudioCodes’ Mediant 800 E-SBC, Mediant 1000 MSBG E-SBC and Mediant 3000 E-SBC Media Gateway

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Notice

This document describes the procedure for integrating the AT&T IP Flexible Reach-MIS/PNT/AVPN SIP Trunking service with Microsoft® Lync Server using the AudioCodes Mediant 800 MSBG-E-SBC, Mediant 1000 MSBG E-SBC and Mediant 3000 E-SBC Media Gateway.

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Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

Note: Throughout this guide, the term *E-SBC device* refers to AudioCodes' Mediant 800 MSBG E-SBC, Mediant 1000 MSBG E-SBC and the Mediant 3000 E-SBC Media Gateway.

Related Documentation

Manual Name
LTRT-26901_SIP_CPE_Release_Notes_Ver._6.2.pdf
LTRT-52306_SIP_CPE_Product_Reference_Manual_Ver_6.2.pdf
LTRT-27001 Mediant 1000 MSBG User's Manual Ver 6.2.pdf
LTRT-40809 Mediant 1000 MSBG Installation Manual Ver 6.2.pdf
LTRT-89710 Mediant 3000 SIP User's Manual Ver 6.2.pdf
LTRT-94708 Mediant 3000 SIP-MGCP-MEGACO Installation Manual Ver 6.2.pdf

1 Introduction

This Configuration Guide describes a sample configuration for a network that uses the AudioCodes Mediant 800 MSBG E-SBC, Mediant 1000 MSBG E–SBC or the Mediant 3000 E-SBC Media Gateway to facilitate a connection between Microsoft Lync 2010 and AT&T's IP Flexible Reach MIS/PNT/AVPN SIP Trunking service for superior voice quality services.

The Mediant 800 MSBG E-SBC is a networking device that combines multiple service functions such as a Media Gateway, Session Border Controller (SBC), Data Router and Firewall, LAN switch, WAN access, Stand Alone Survivability (SAS) and an integrated general-purpose server.

The Mediant 1000 MSBG E-SBC is all-in-one multi-service access solution products for Service Providers (SME's) offering managed services and distributed Enterprises seeking integrated services. This multi-service business gateway is designed to provide converged Voice & Data services for business customers at wire speed, while maintaining SLA parameters for superior voice quality.

The Mediant 1000 MSBG E-SBC is based on AudioCodes' VoIPerfect Media Gateway technology, combined with Enterprise class Session Border Controller, Data & Voice security elements, Data Routing, LAN Switching and WAN Access. These services allow smooth connectivity to cloud services, while providing protection to the end customer.

The Mediant 3000 E-SBC Media Gateway is a High Availability VoIP Gateway and Enterprise Class SBC for medium and large enterprises.

Note: The scope of this document does not cover security aspects for connecting the SIP Trunk to the Microsoft Lync environment. Security measures should be implemented in accordance with your organization's security policies. For basic security guidelines, see the 'AudioCodes Security Guidelines'.

Reader's Notes

2 Testing Considerations

Note the following special considerations for the AT&T test environment:

- Fax was not tested and is not supported.
- G.711 U-law is the only codec supported for this application.
- Music on hold does not work on IP Flexible Reach telephone numbers (TNs) that are served by the AT&T legacy SBC local PSTN footprint.
- Voice mail should work; however was not tested during the certification.
- Emergency 911/E911 Services Limitations and Restrictions - Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer Configuration guide, will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is Customer's responsibility to ensure proper operation with its equipment/software vendor.
- The transfer calls were tested with REFER and AudioCodes terminates the messages and supports the call flow directly.

  Transfer calls were tested with the Microsoft Lync environment pre-configured to send REFER messages towards the SIP trunk. As REFER messages were sent from the Microsoft Lync environment, the messages were processed by AudioCodes directly and not forwarded on to the AT&T IP Flexible Reach trunk. Direct handling of these messages is a proven AudioCodes feature and a proven interworking functionality between AudioCodes and Microsoft Lync 2010.
- While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when the E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at http://new.serviceguide.att.com <http://new.serviceguide.att.com/>. Such circumstances include; however are not limited to relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions.

Reader's Notes

3 Scenario Overview

The configuration scenario described in this document includes the following setup:

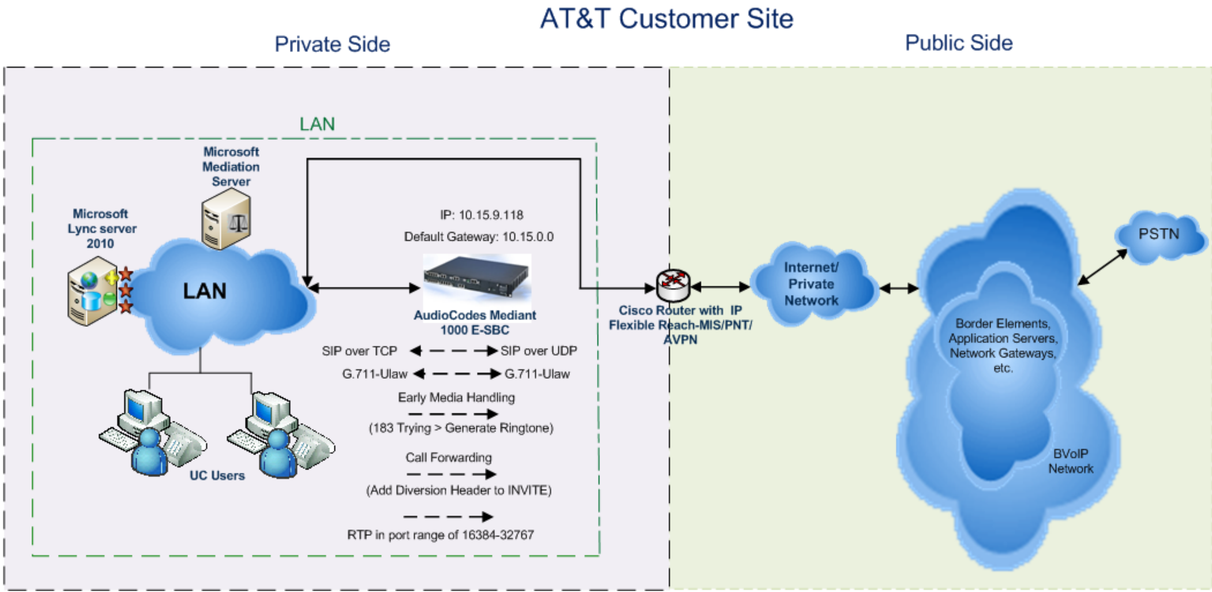
- An Enterprise has a deployed Microsoft® Lync server 2010 in its private network for enhanced communication within the company.
- The enterprise decides to offer its employees Enterprise voice and to connect the company to the PSTN network using the Flexible Reach SIP Trunking service.

The setup requirements are as follows:

- While the Microsoft® Lync Server 2010 environment is located on the Enterprise's Local Area Network (LAN), the Flexible Reach SIP Trunks are located on the WAN.
- Microsoft® Lync Server 2010 works with the TCP transport type, while the Flexible Reach SIP trunk works on the SIP over UDP transport type.
- Both Microsoft® Lync Server 2010 and Flexible Reach SIP Trunk support the G.711-Ulaw coder type.
- Support for early media handling
- Support for call forwarding

The figure below illustrates an overview of the configuration scenario.

Figure 3-1: Scenario Overview



Reader's Notes

4 Configuring Microsoft Lync Server 2010

This section describes how to configure the Microsoft Lync Server 2010 to operate with the E-SBC device. This section describes the following procedures:

1. Configuring the E-SBC device as an ‘ IP/PSTN Gateway’. See Section 4.1 on page 16.
2. Associating the ‘IP/PSTN Gateway’ with the Mediation Server. See Section 4.2 on page 21.
3. Configuring a ‘Route’ to utilize the SIP trunk connected to the E-SBC device. See Section 4.3 on page 27.

> **Note:** Dial Plans, Voice Policies, and PSTN usages are also necessary for enterprise voice deployment; however, they are beyond the scope of this document.

4.1 Configuring the AudioCodes E-SBC device as a 'IP/PSTN Gateway'

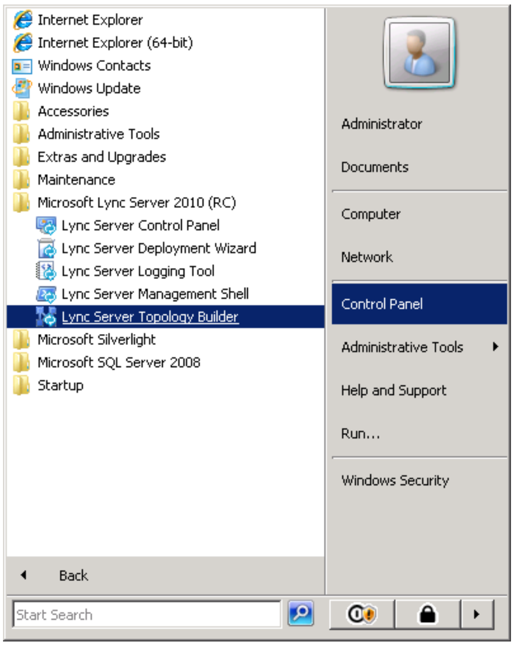
This section describes how to configure the E-SBC device as an IP/PSTN Gateway.

Note: The Microsoft Lync Topology Builder interface dialogs refer to the E-SBC device as an 'IP/PSTN gateway' or 'PSTN gateway'.

➢ To configure the E-SBC device as a IP/PSTN Gateway and associating it with the Mediation Server:

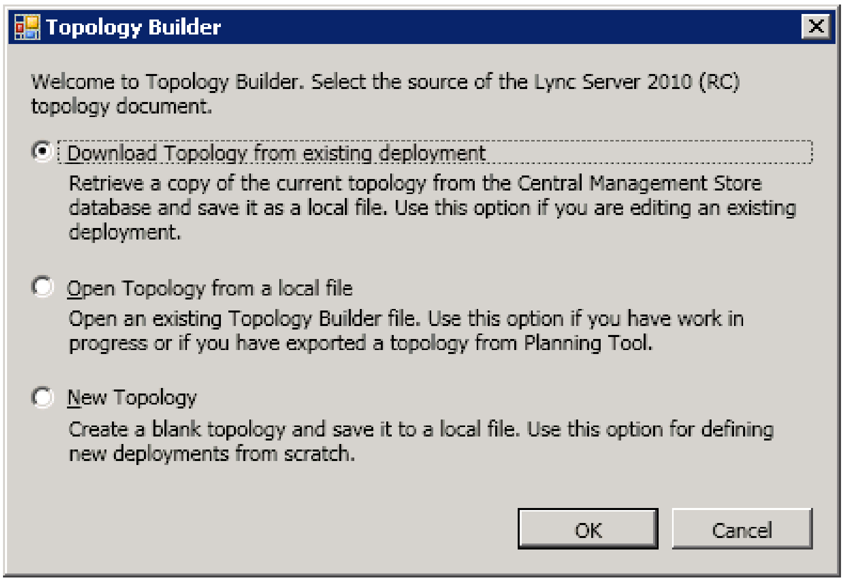
1. On the server where the Topology Builder is located, start the Microsoft Lync Server 2010 **Topology Builder**: Click **Start**, select **All Programs**, then select **Lync Server Topology Builder**.

Figure 4-1: Starting the Lync Server Topology Builder



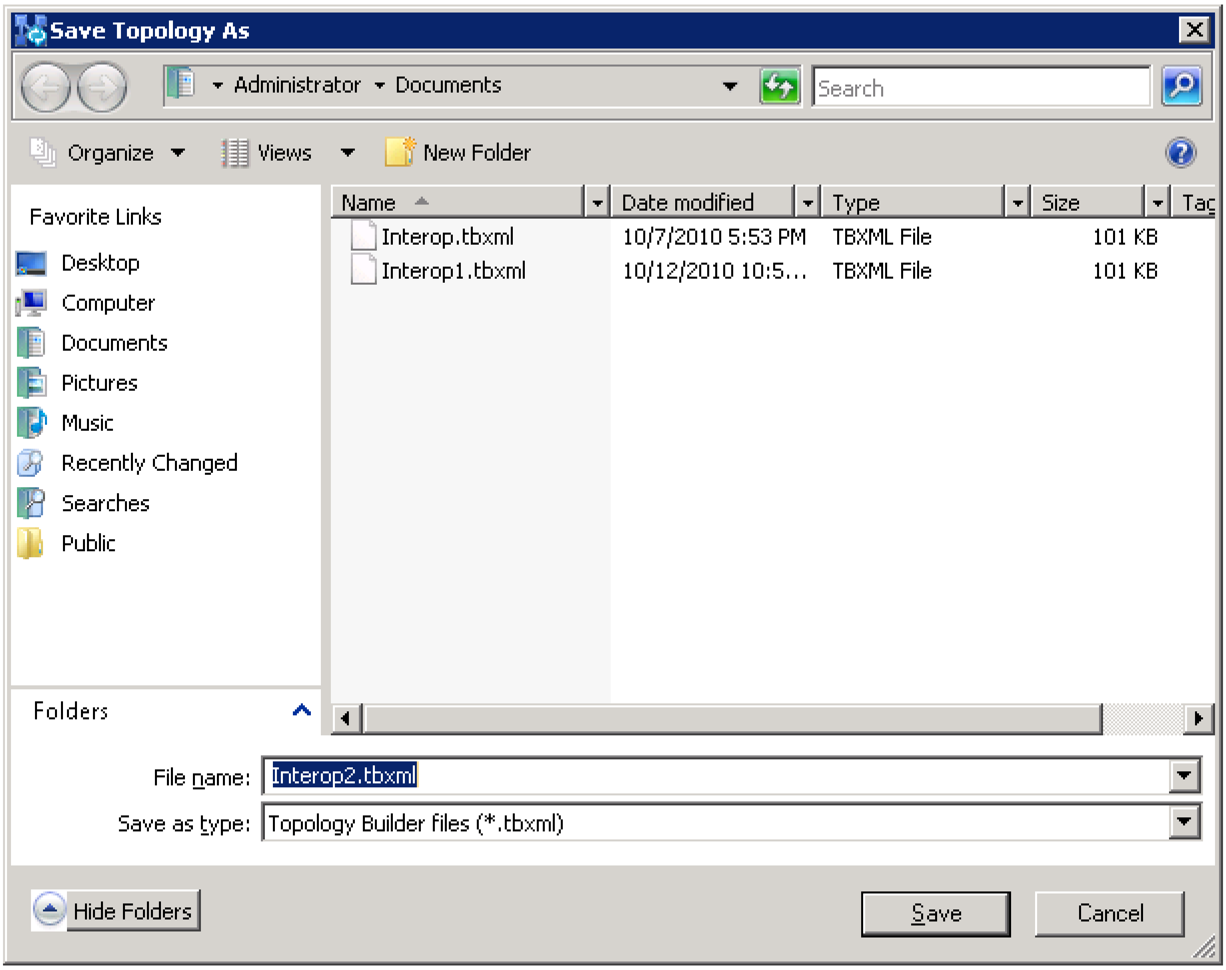
The following screen is displayed:

Figure 4-2: Topology Builder Options



2. Choose ‘Download Topology from the existing deployment and click **OK**.
You are prompted to save the Topology which you have downloaded.

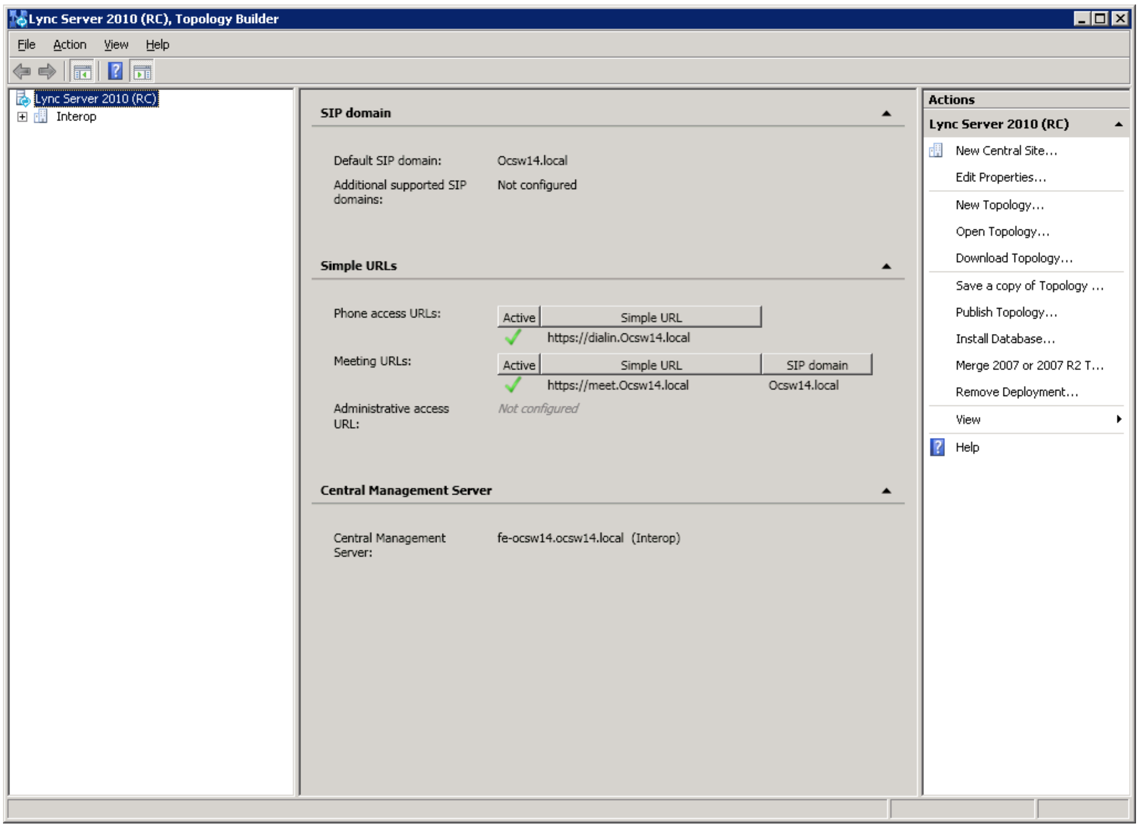
Figure 4-3: Save Topology



3. Enter new **File Name** and **Save** – this action enables you to rollback from any changes you make during the installation.

   The Topology Builder screen with the topology downloaded is displayed.

Figure 4-4: Downloaded Topology



4. Expand the Site; right-click on the IP/PSTN Gateway and choose 'New IP/PSTN Gateway'.

Figure 4-5: New IP/PSTN Gateway

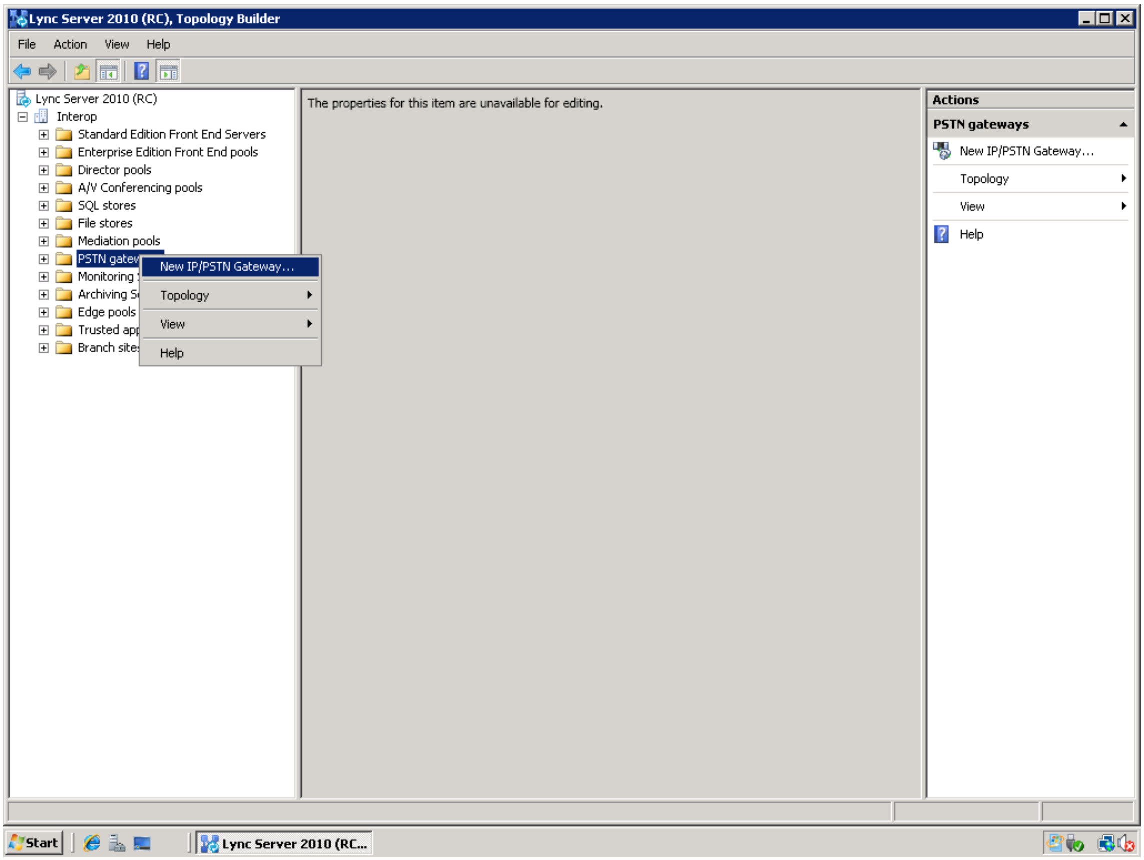
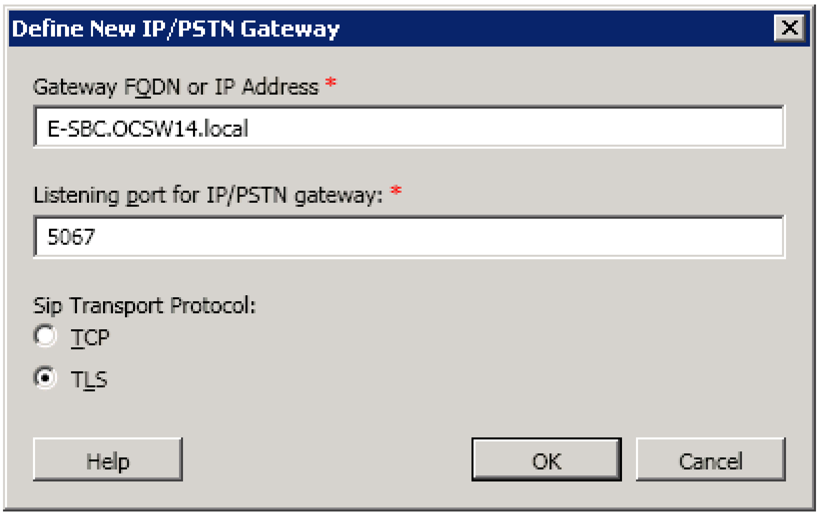


Figure 4-6: Define New IP/PSTN Gateway

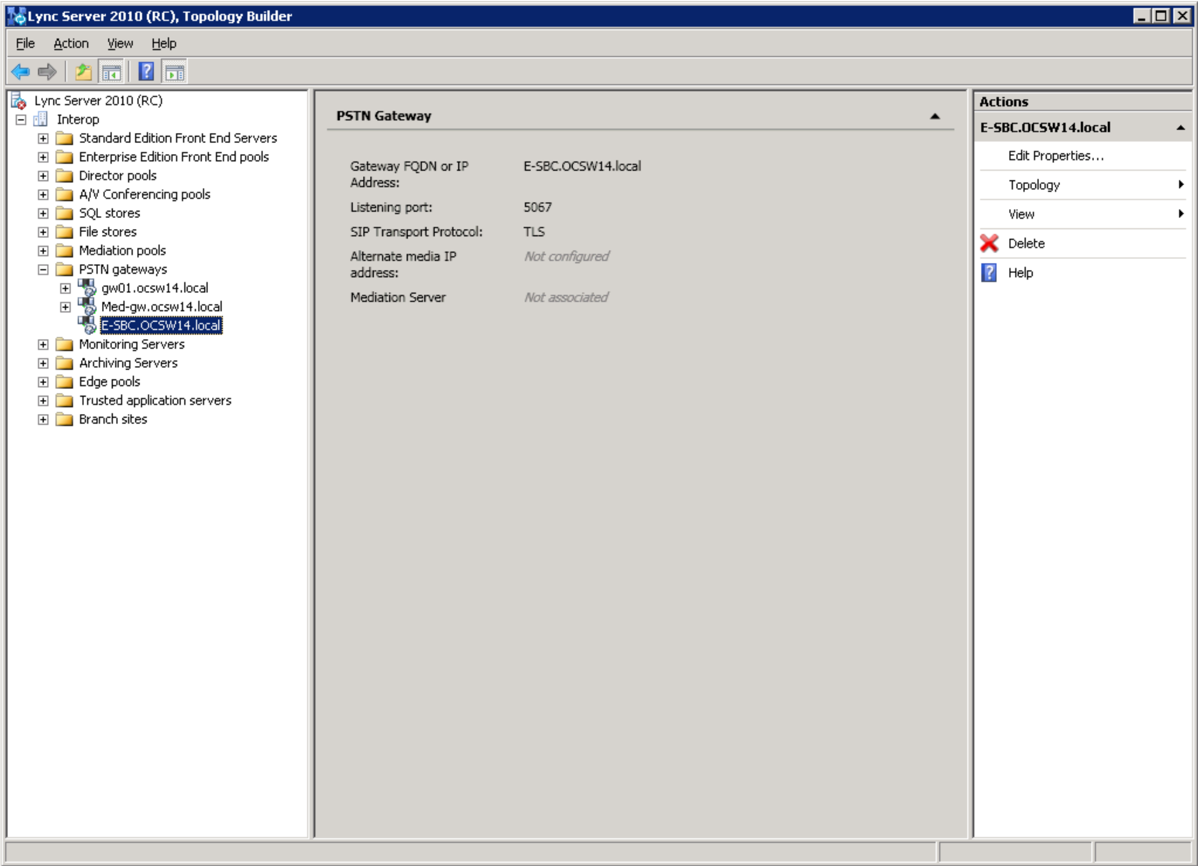


5. Enter the FQDN of the E-SBC device (i.e. 'E-SBC.OCSW14.local') and click **OK**.

   Note that the listening port for the Gateway is '5067' and the transport type is 'TLS'. In certification testing for the AT&T IP Flexible Reach SIP trunk, listening port 5060 was used with transport protocol 'TCP'.

   The E-SBC device is now added as an 'IP/PSTN Gateway'.

Figure 4-7: IP/PSTN Gateway



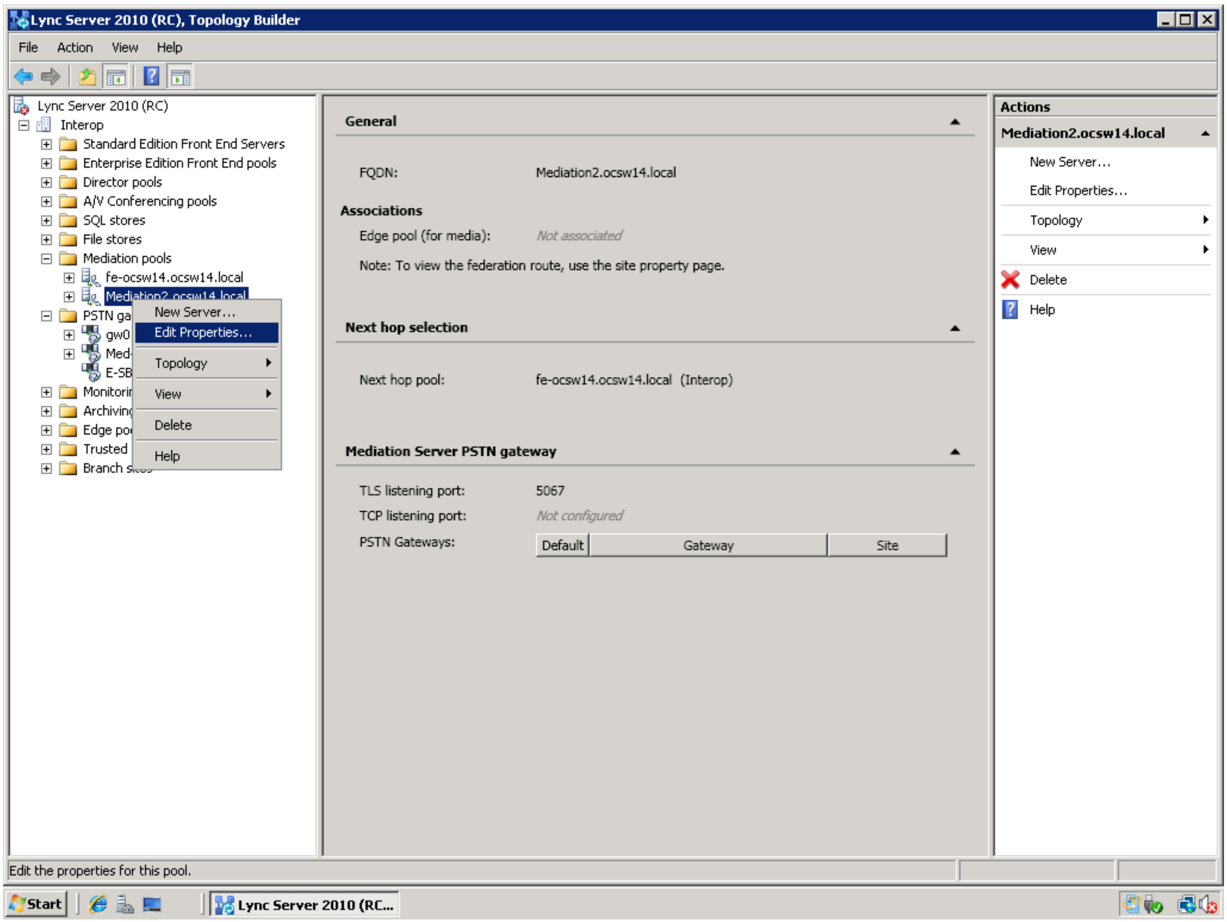
4.2 Associating the 'IP/PSTN Gateway' with the Mediation Server

This section describes how to associate the 'IP/PSTN Gateway' (E-SBC device) with the Mediation Server.

➢ **To associate the IP/PSTN Gateway with the Mediation Server:**

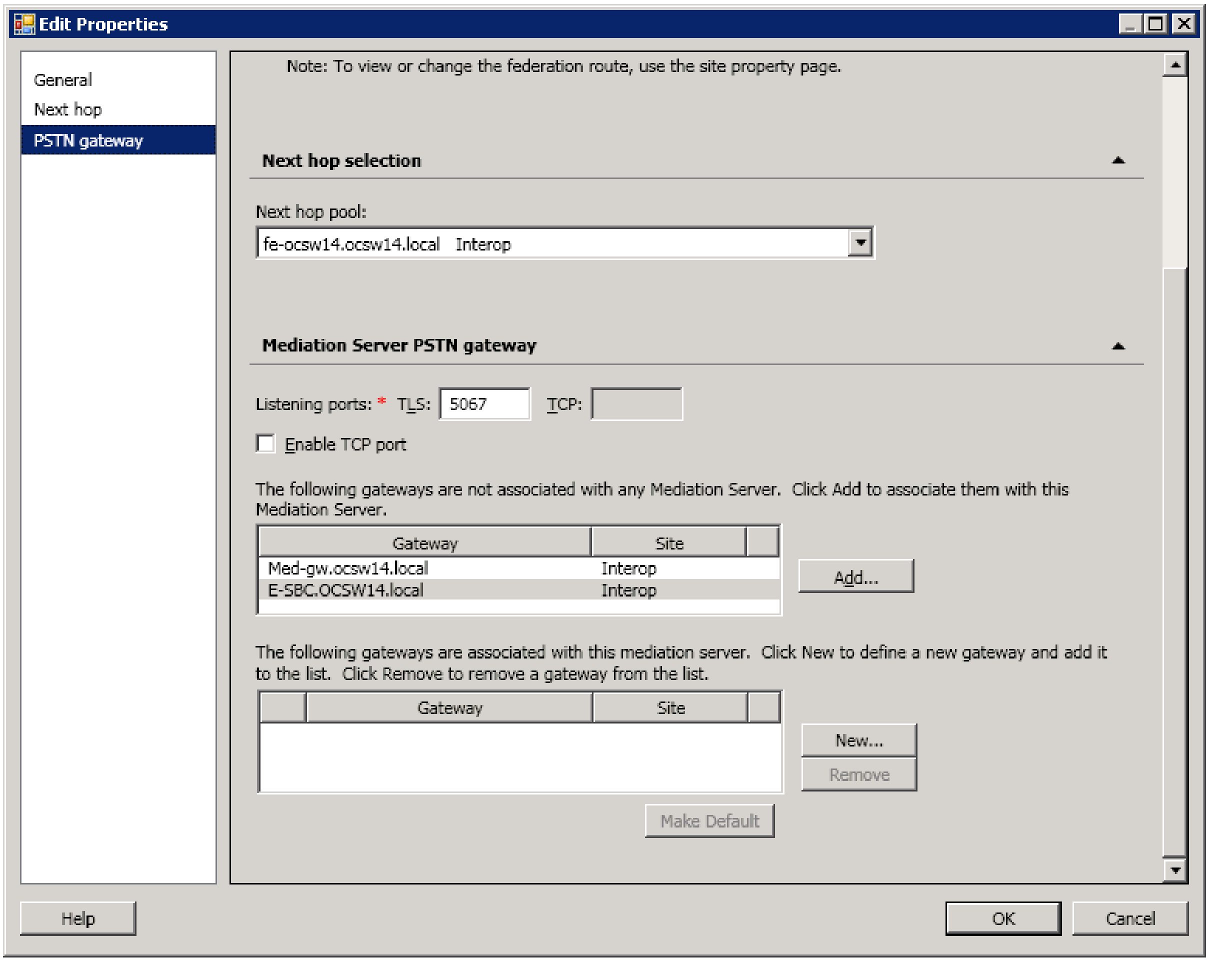
1. Right-click on the **Mediation Server** to use with the IP/PSTN Gateway (i.e. Mediation2.OCSW14.local) and choose **Edit Properties**.

Figure 4-8: Associating Mediation Server with IP/PSTN Gateway



The following screen is displayed:

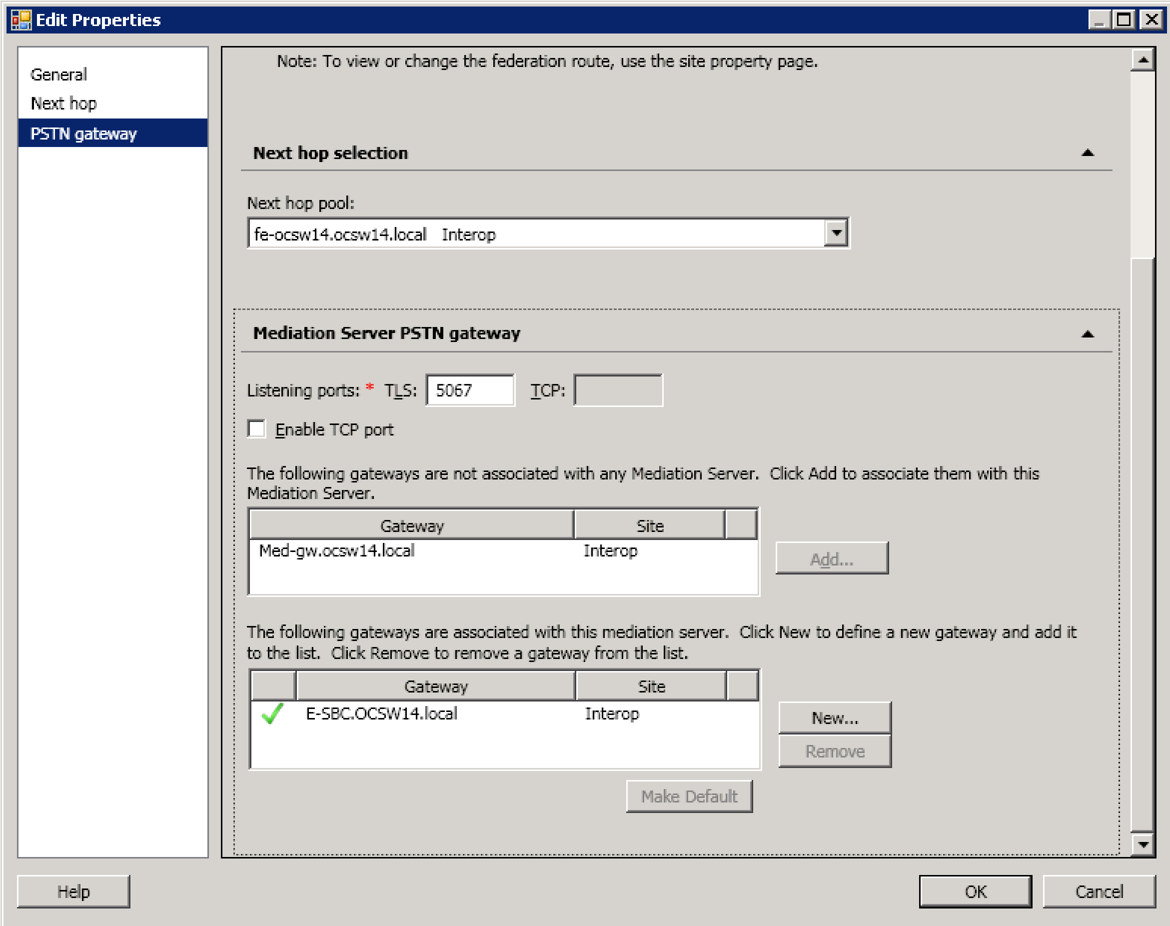
Figure 4-9: Before Associating IP/PSTN Gateway to a Mediation Server



2. In the top-left corner, choose **PSTN gateway** and in the Mediation Server PSTN gateway pane, mark the E-SBC device that is designated as the IP/PSTN gateway (i.e. 'E-SBC.OCSW14.local') and click **Add** to associate it with this Mediation Server.

   Note that there are two sub-panes, one including a list of IP/PSTN gateways not associated with the Mediation Server and one including a list of IP/PSTN gateways associated with the Mediation server.

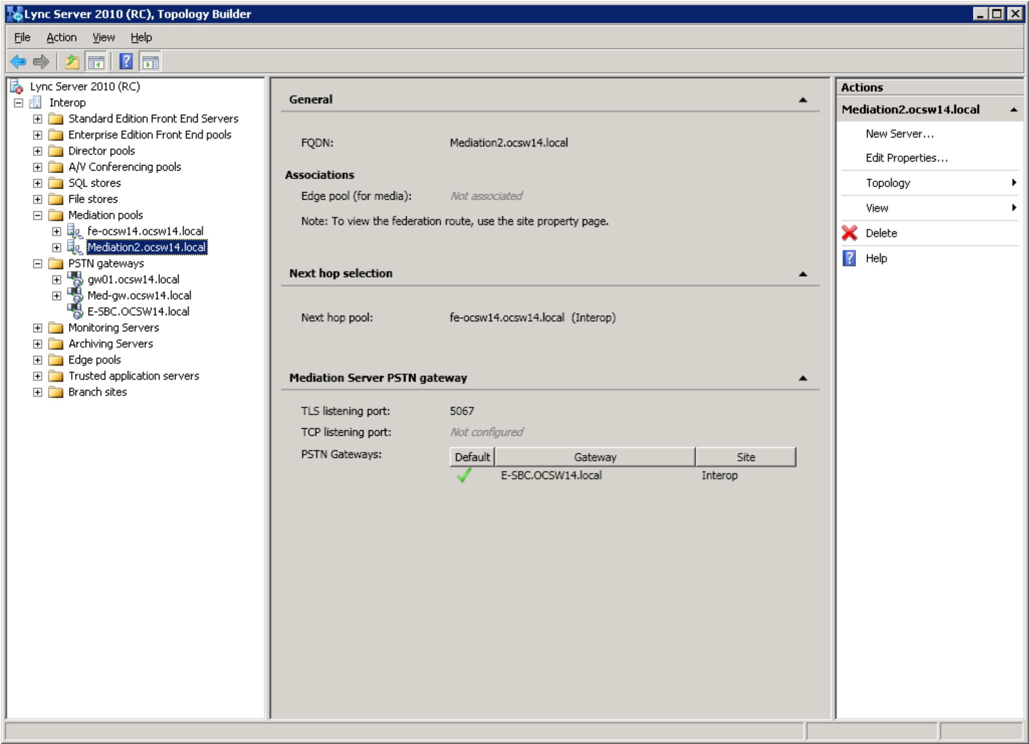
Figure 4-10: After Associating IP/PSTN Gateway to Mediation Server



In the Mediation Server PSTN gateway pane, the IP/PSTN Gateway that you associated with the Mediation Server is displayed with an adjacent Green ✓.

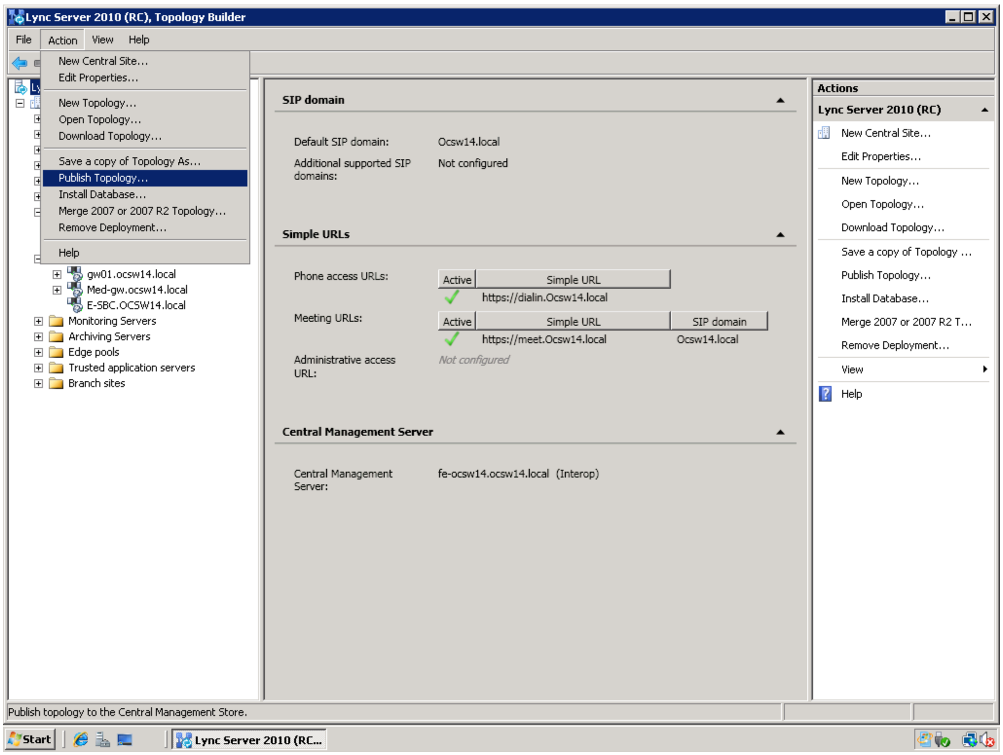
3. Click **OK**.

Figure 4-11: Media Server PSTN Gateway Association Properties



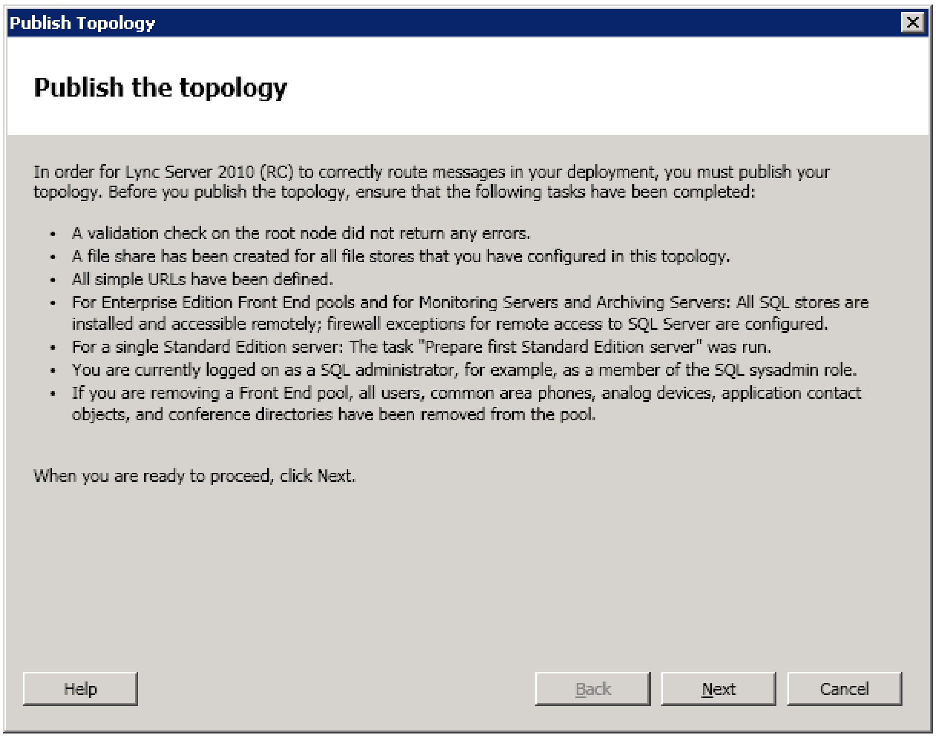
4. In the Lync Server main menu, choose **Action > Publish Topology**.

Figure 4-12: Publishing Topology



The Publish Topology screen is displayed.

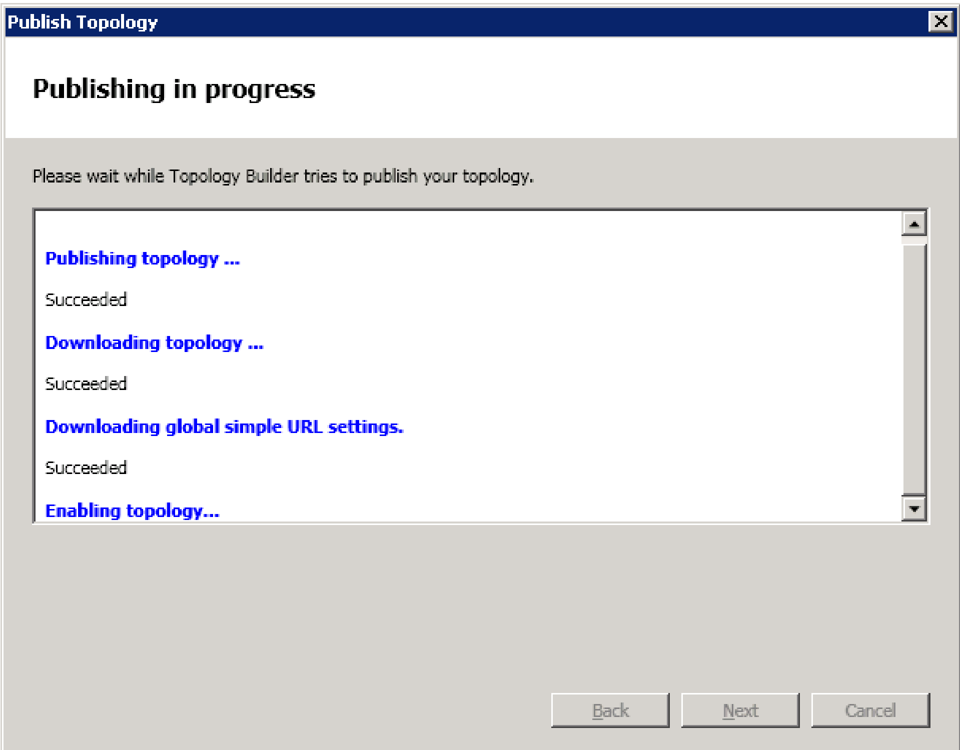
Figure 4-13:Publish Topology Confirmation



5. Click **Next**.

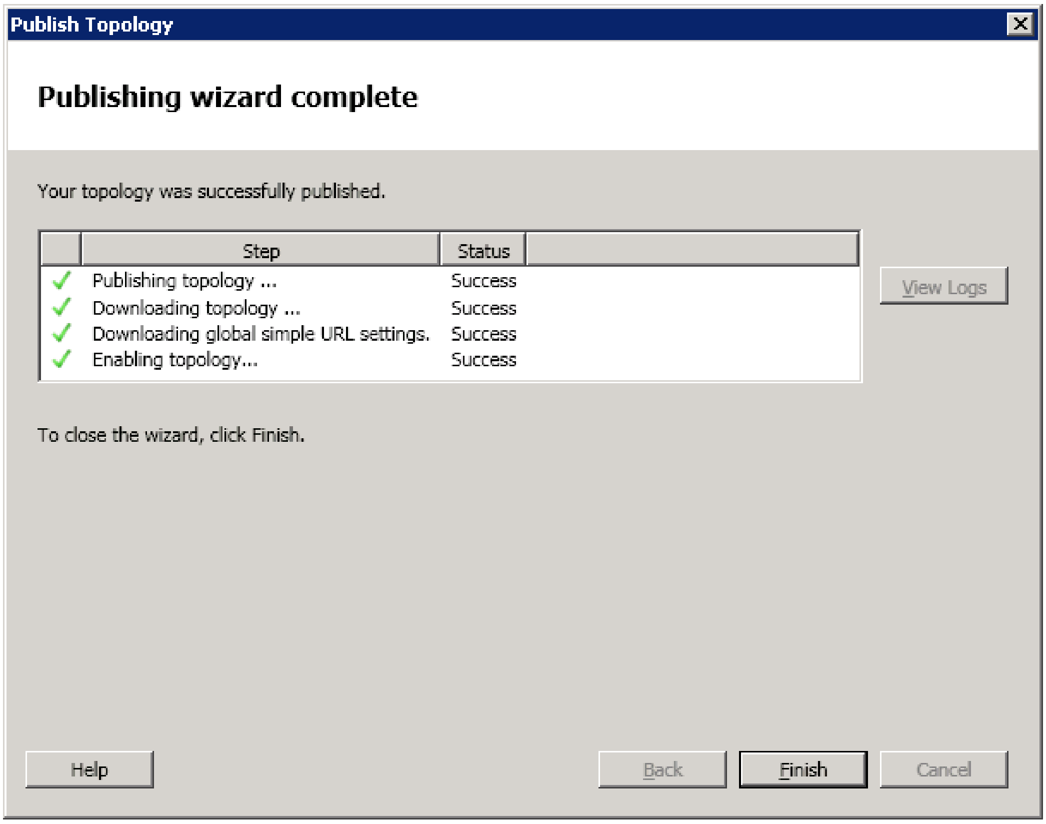
The Topology Builder attempts to publish your topology.

Figure 4-14: Publish Topology Confirmation screen



Wait until the publish topology process has ended successfully.

Figure 4-15: Publish Topology Successfully Completed



6. Click **Finish**.

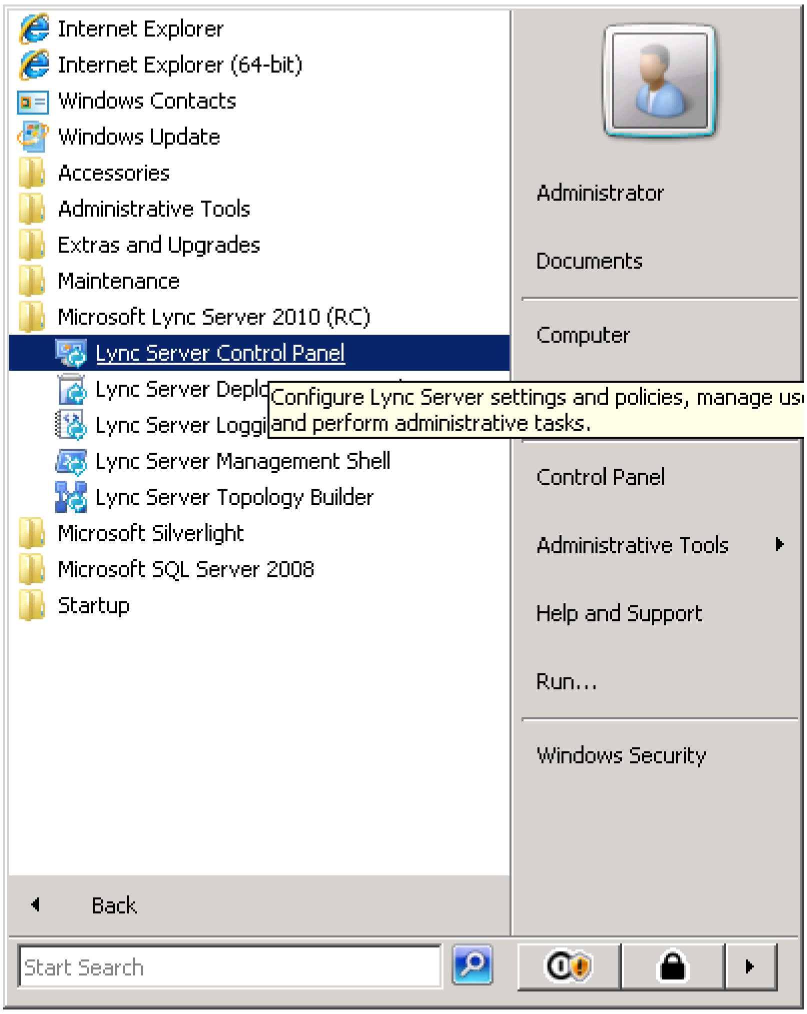
4.3 Configuring the 'Route' on the Lync Server 2010

This section describes how to configure a 'Route' on the Lync server and associates it with the IP/PSTN gateway.

➢ **To configure the 'route' on the Lync server:**

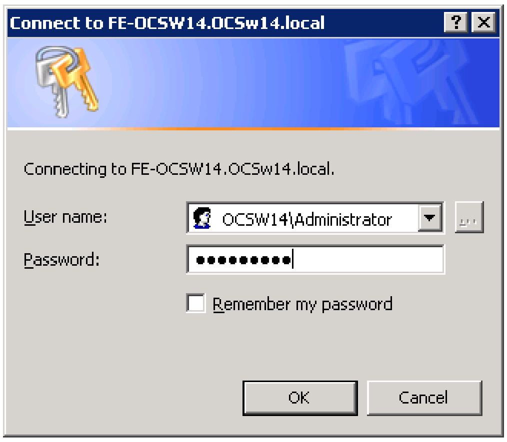
1. Open the Communication Server Control Panel (CSCP), click **Start**, select **All Programs**, and select **Lync Server Control Panel**.

Figure 4-16: Lync Server Control Panel



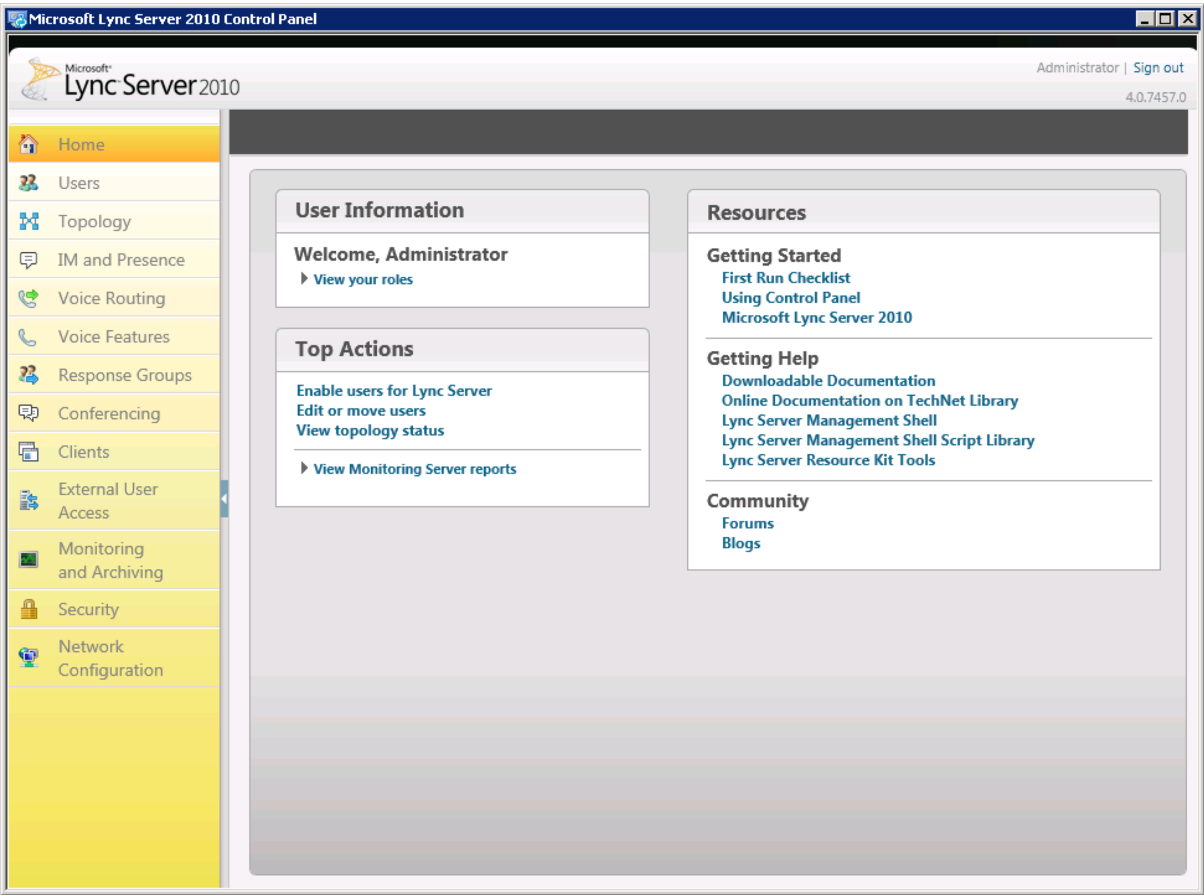
2. You are prompted for credentials; enter your domain username and password.

Figure 4-17: Lync Server Credentials



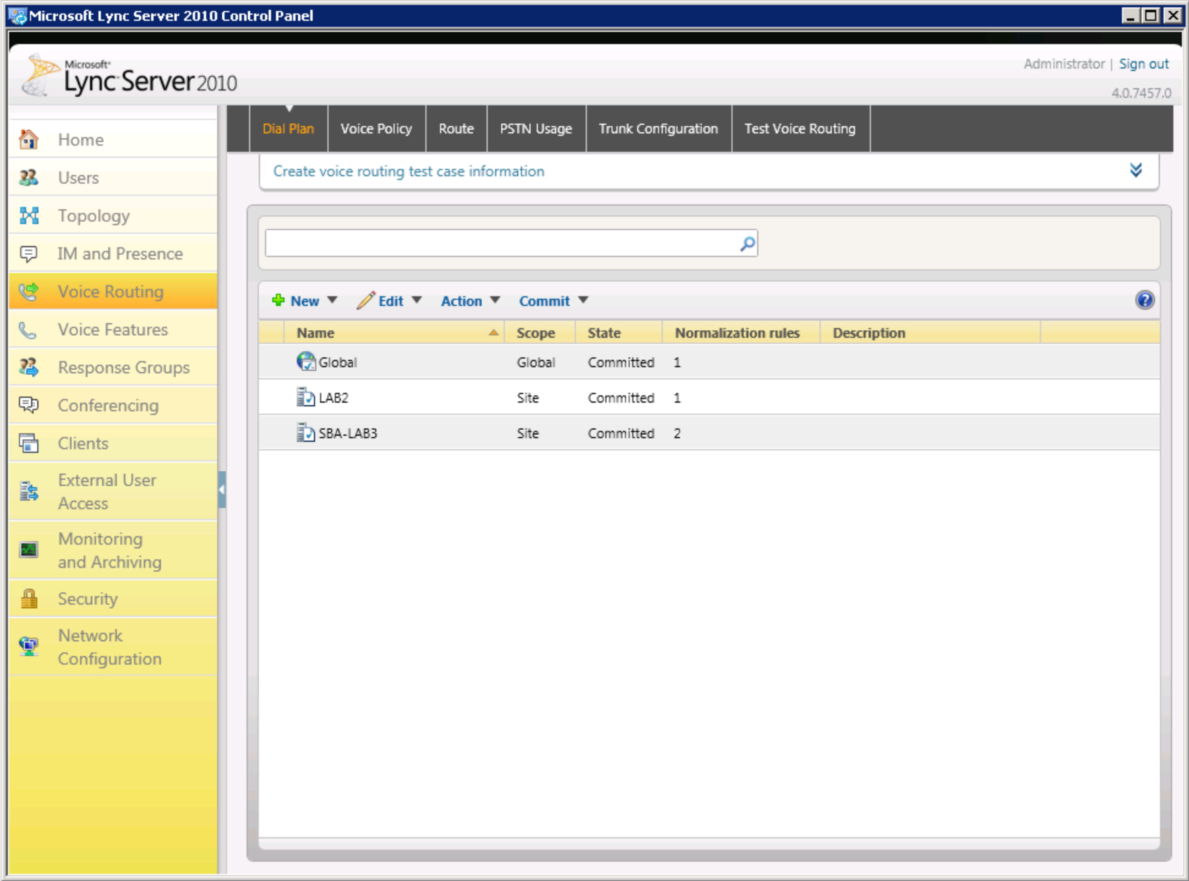
The CSCP Home page is displayed.

Figure 4-18: CSCP Home page



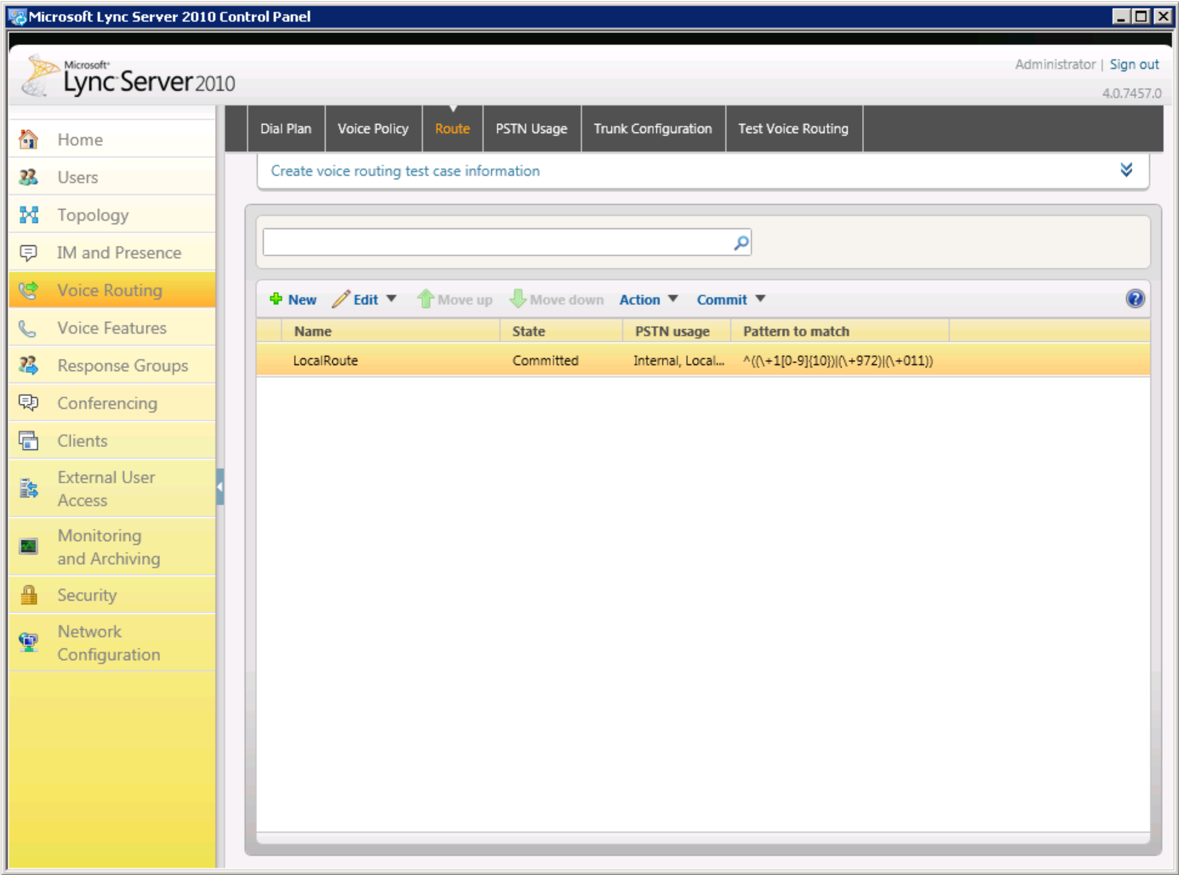
3. In the Navigation pane, select the **'Voice Routing'** option.

Figure 4-19: Voice Routing Option



4. In the Voice Routing menu at the top of the page, select the **Route** option.

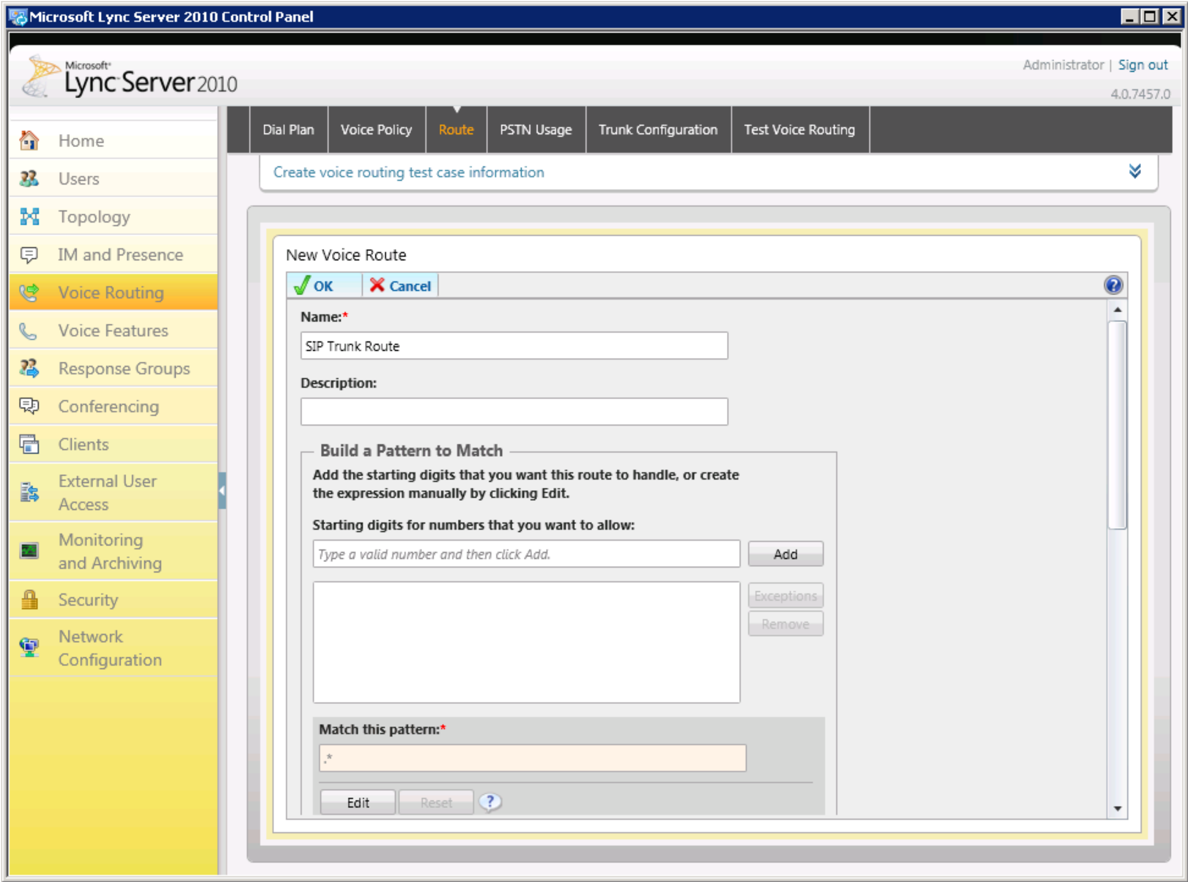
Figure 4-20: Route Option



5. In the content area toolbar, click 

.
6. In the Build a Pattern to Match pane, fill in a Name for this route (i.e SIP Trunk Route) and a Pattern to Match for the phone numbers you wish this route to handle. In this example, the pattern to match is '*', which implies "to match all numbers".
7. Click Add.

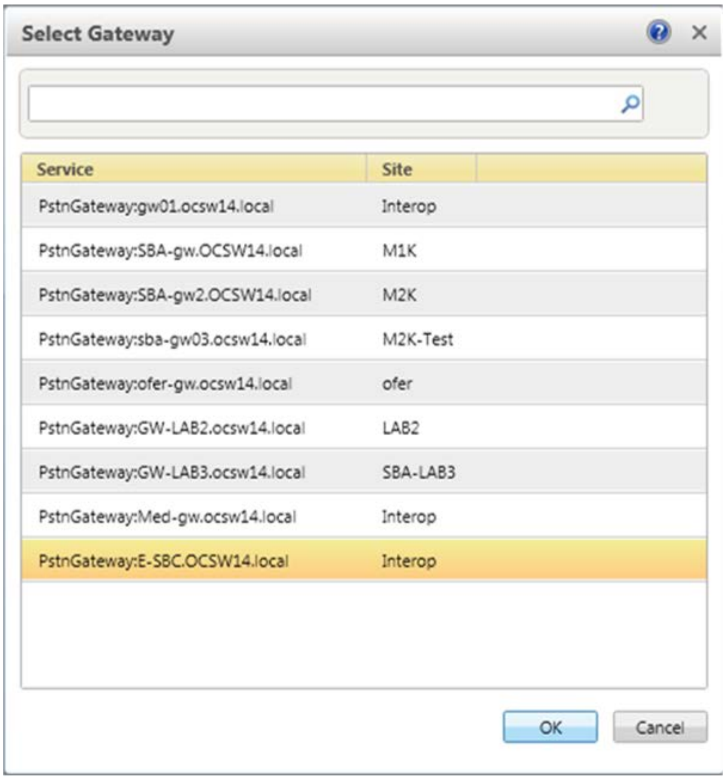
Figure 4-21: Adding New Voice Route



8. Associate the route with the IP/PSTN gateway you created above; scroll down to the Associated Gateways pane and click **Add**.

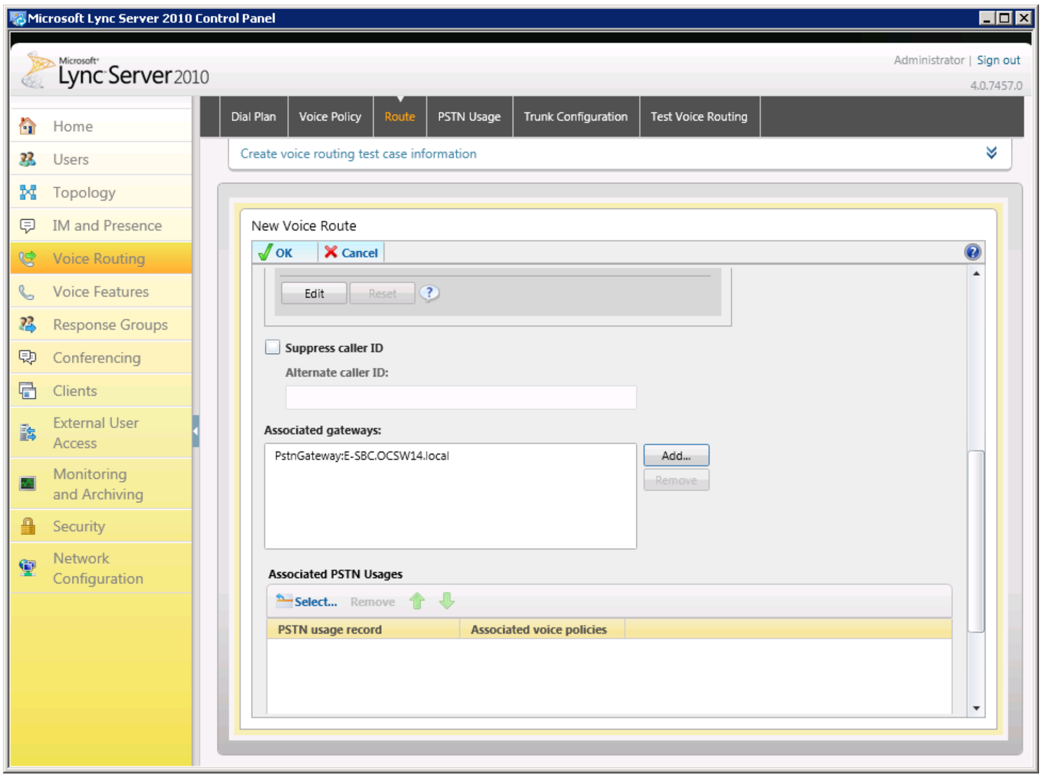
   A list of all the deployed Gateways is displayed.

Figure 4-22: List of Deployed Gateways



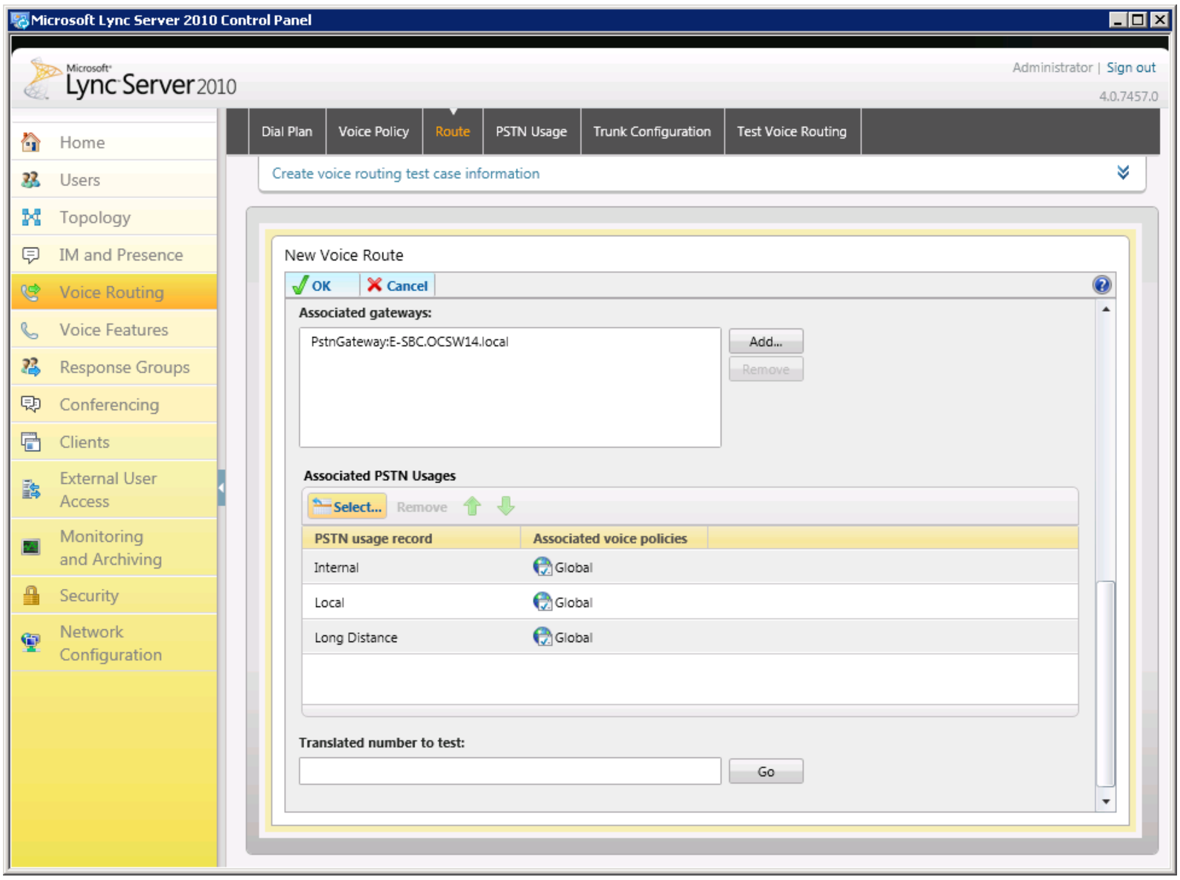
9. Select the IP/PSTN Gateway you created above and click **OK**.

Figure 4-23: Selecting the PSTN Gateway



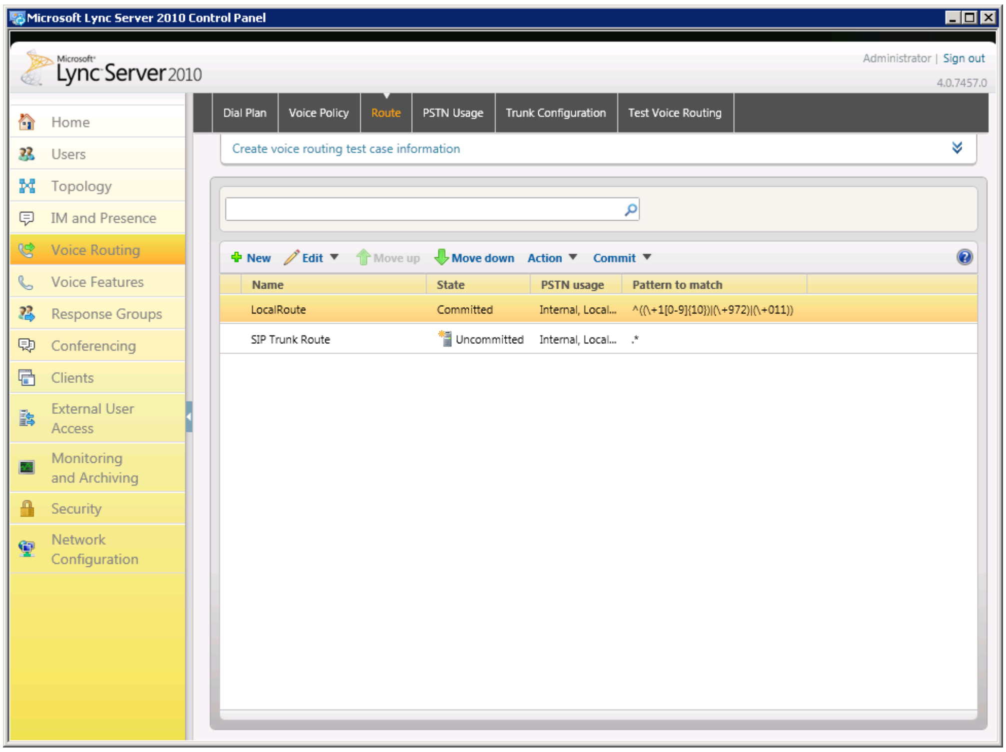
10. Associate a PSTN Usage to this route. In the **Associated PSTN Usages** toolbar, click **Select** and add the associated PSTN Usage.

Figure 4-24: Associating PSTN Usage to PSTN Gateway



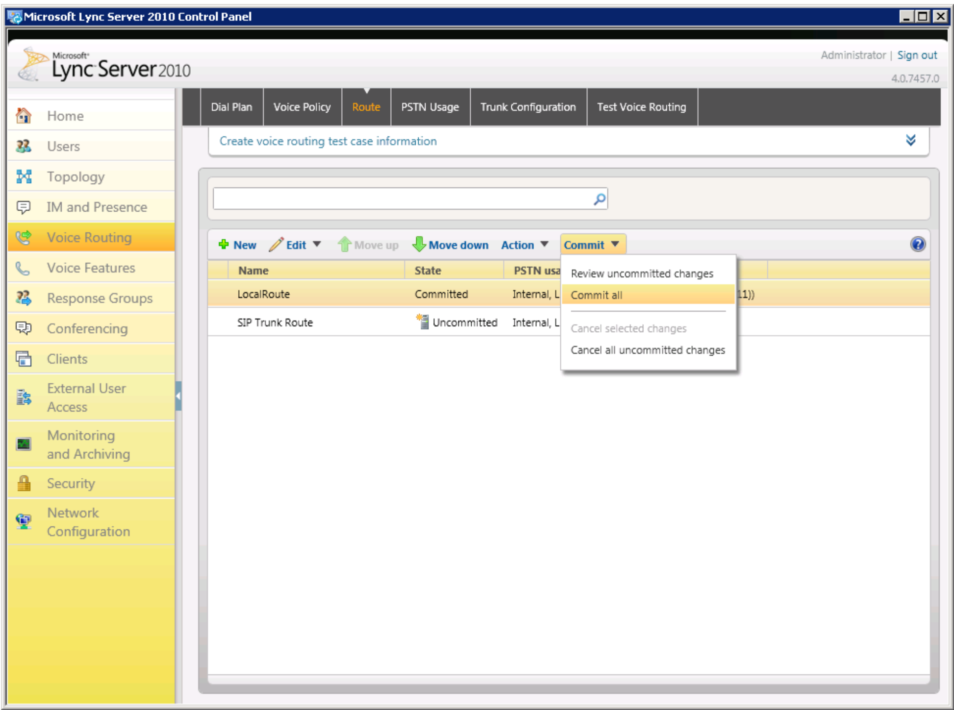
11. Click the **OK** button in the toolbar at the top of the New Voice Route pane.

Figure 4-25: Confirmation of New Voice Route



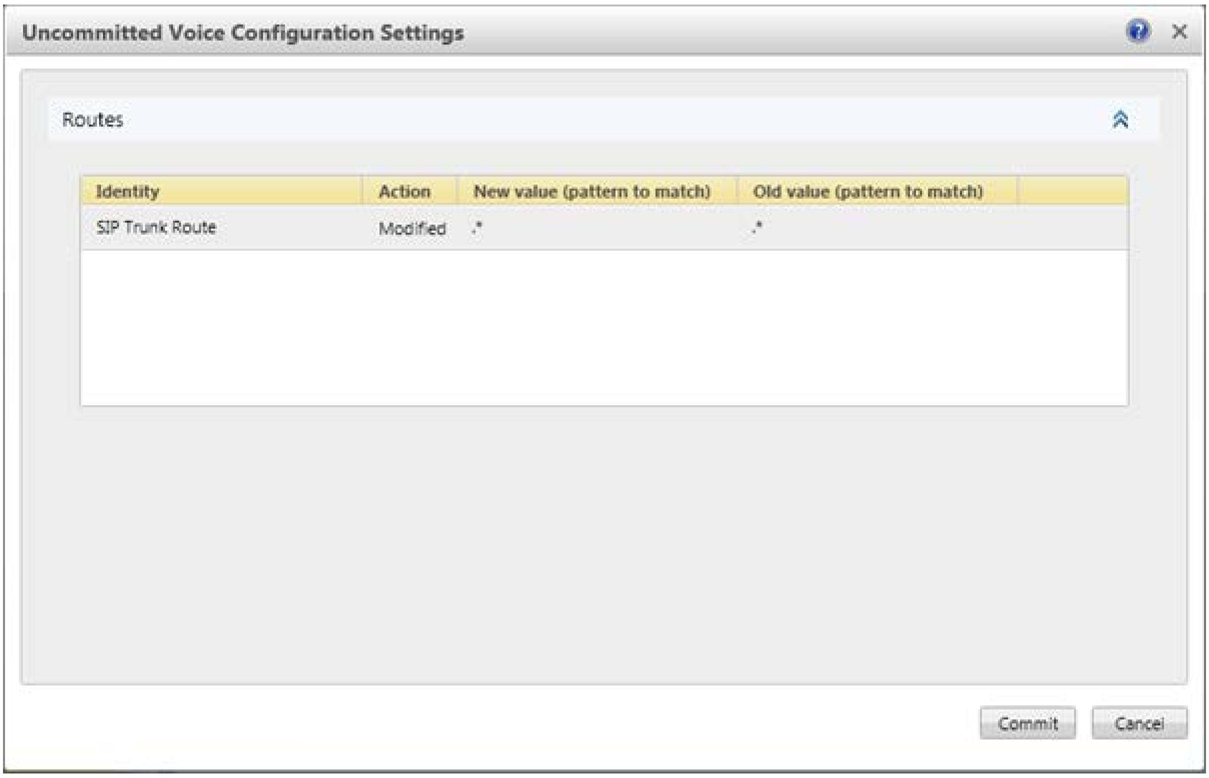
12. In the Content area Toolbar, click on the arrow adjacent to the **Commit** button; a drop-down menu is displayed; select the 'Commit All' option.

Figure 4-26: Committing Voice Routes



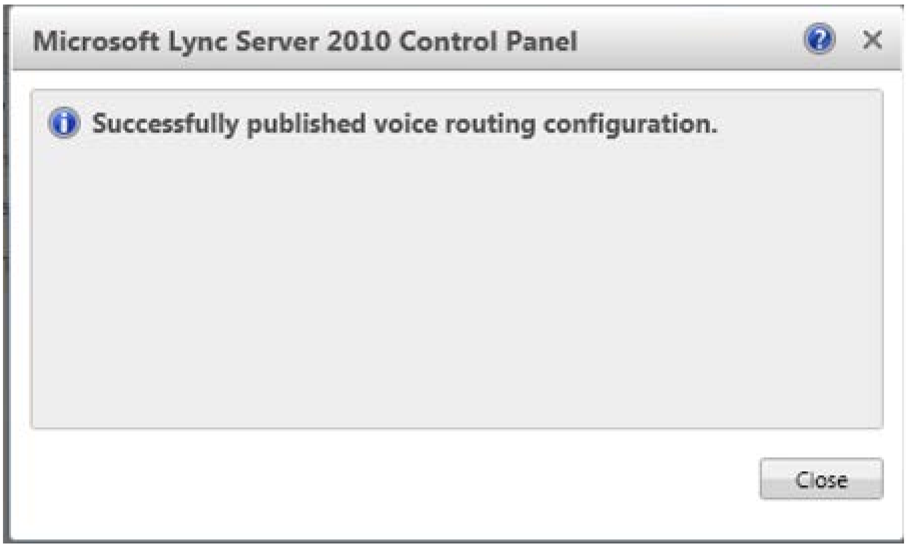
13. In the Uncommitted Voice Configuration Settings window, click **Commit**.

Figure 4-27: Uncommitted Voice Configuration Settings



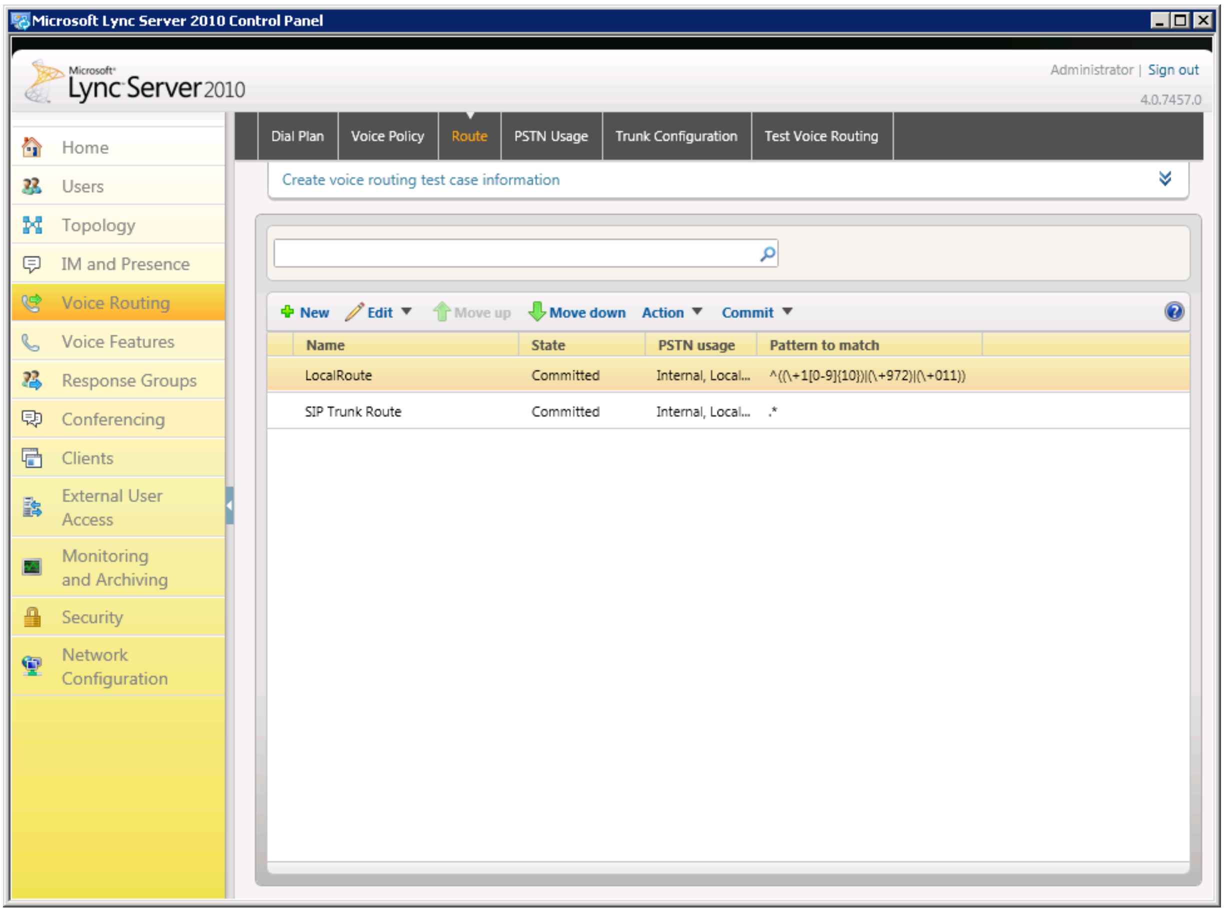
14. A message is displayed, confirming a successful voice routing configuration; in the **Microsoft Lync Server 2010 Control Panel** prompt, click **Close**.

Figure 4-28: Voice Routing Configuration Confirmation



The new committed Route is now displayed in the Voice Routing screen.

Figure 4-29: Voice Routing Screen Displaying Committed Routes



5 Configuring E-SBC Device

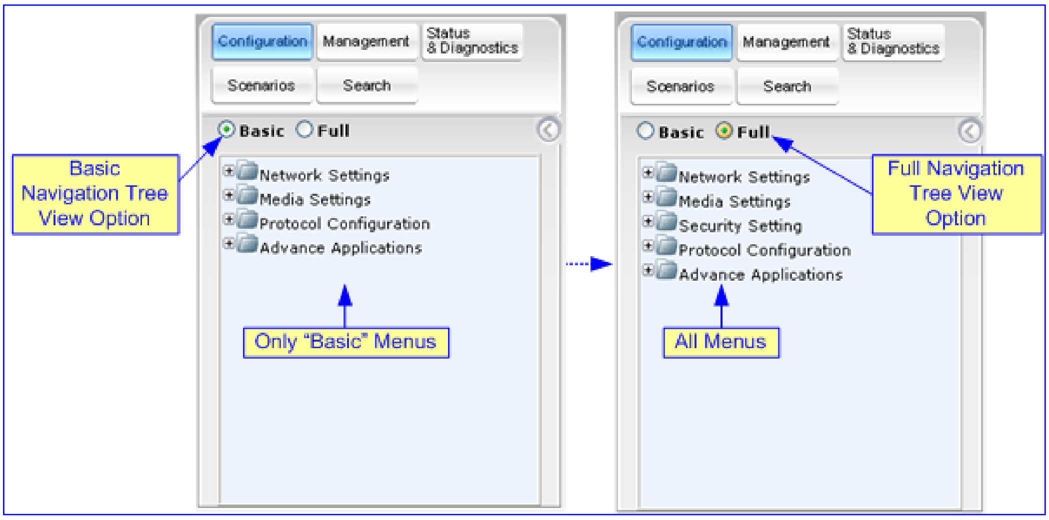
This section provides step-by-step procedures for configuring the E-SBC device.

The following describes the steps required to configure the E-SBC device :

- **Step 1**: Configure IP Addresses. See section 5.1 on page 38.
- **Step 2**: Enable the SBC Capabilities. See section 5.2 on page 39.
- **Step 3**: Configure the Number of Media Channels. See section 5.3 on page 40.
- **Step 4**: Configure the Proxy Sets. See section 5.4 on page 41.
- **Step 5**: Configure the IP Groups. See section 5.5 on page 43.
- **Step 6**: Configure the Voice Coders. See section 5.6 on page 45.
- **Step 7**: Define Silence Suppression and Comfort Noise. See section 5.6.1 on page 46.
- **Step 8**: Configure IP Profile Settings. See section 5.7 on page 47.
- **Step 9**: Configure IP-to-IP Routing Setup. See section 5.8 on page 49.
- **Step 10**: Configure Number Manipulation. See section 5.9 on page 52.
- **Step 11**: Configuring IP Profile for Call Forwarding. See section 5.10 on page 57.
- **Step 12**: Configuring SIP General Parameters. See section 5.11 on page 60.
- **Step 13**: Defining Reasons for Alternative Routing. See section 5.12 on page 62.

The procedures described in this section are performed using the E-SBC devices' Web-based management tool (i.e., embedded Web server). Before you begin configuring the E-SBC device, ensure that the Web interface's Navigation tree is in full menu display mode (i.e., the **Full** option on the Navigation bar is selected), as displayed below:

Figure 5-1: Web Interface Showing Basic/Full Navigation Tree Display



5.1 Step 1: Configure IP Addresses

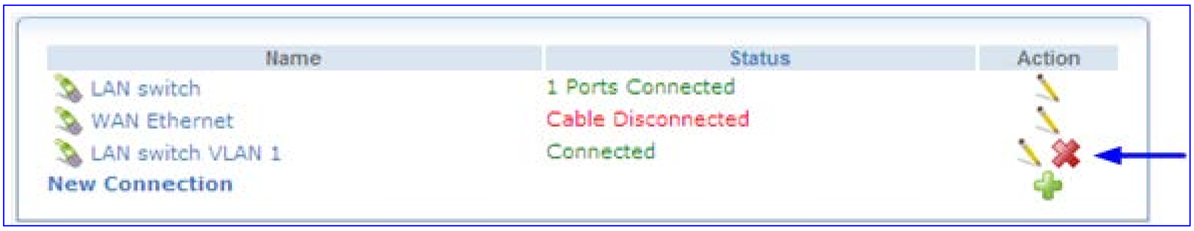
This section describes how to configure IP addresses when a single LAN interface is used to connect to the Flexible Reach SIP Trunk. In this configuration, the internal data-routing capabilities of the E-SBC device are not used. As a consequence, you must disable the internal data-routing interface as described in the procedure below.

Note: When operating in LAN VoIP-only mode, do not use the E-SBC device's WAN port.

➢ **To operate the E-SBC device as a LAN VoIP gateway only:**

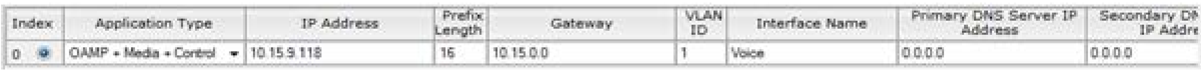
1. Disconnect the network cable from the WAN port and then connect one of the E-SBC device's LAN ports to the network.
2. Disable or remove the data-routing IP network interface:
   - Access the 'Connections' page (Configuration tab > Data menu > Data System > Connections).
   - Delete the "LAN Switch VLAN 1" connection by clicking the corresponding Remove button, and then clicking OK to confirm deletion.

Figure 5-2: Removing Data-Routing Connection Interface



3. Configure VoIP IP network interfaces in the 'Multiple Interface' table (**Configuration** tab > **VoIP** menu > **Network** > **IP Settings**).
   - In the 'Multiple Interface' table, define a single IP network interface for application types "OAMP + Media + Control".

Figure 5-3: Multiple Interface Table



Index	Application Type	IP Address	Prefix Length	Gateway	VLAN ID	Interface Name	Primary DNS Server IP Address	Secondary DN IP Addre
0	OAMP + Media + Control	10.15.9.118	16	10.15.0.0	1	Voice	0.0.0.0	0.0.0.0

   - Click **OK** to save settings.

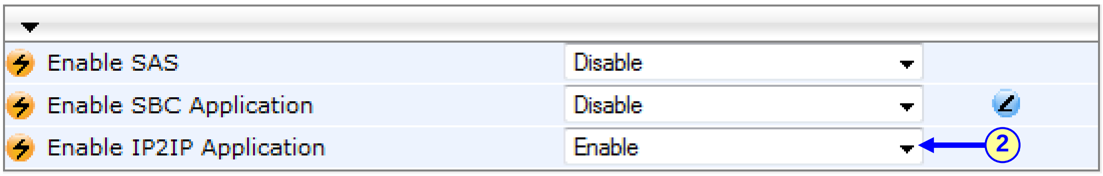
5.2 Step 2: Enable the SIP SBC Application Mode

This step describes how to enable the gateway-SBC devices' SIP SBC application mode.

➢ To enable the SIP SBC application mode:

1. Open the 'Application Enabling' page (**Configuration** tab > **VoIP** menu > **Applications Enabling** > **Applications Enabling**).

Figure 5-4: Application Enabling



2. From the 'Enable IP2IP Application' drop-down list, select "Enable".

   Reset with BURN to FLASH is required.

Note: To enable the IP2IP capabilities on the AudioCodes device, your device must be loaded with the feature key that includes the IP2IP feature and also the E-SBC device must be running SIP version 6.2 or later.

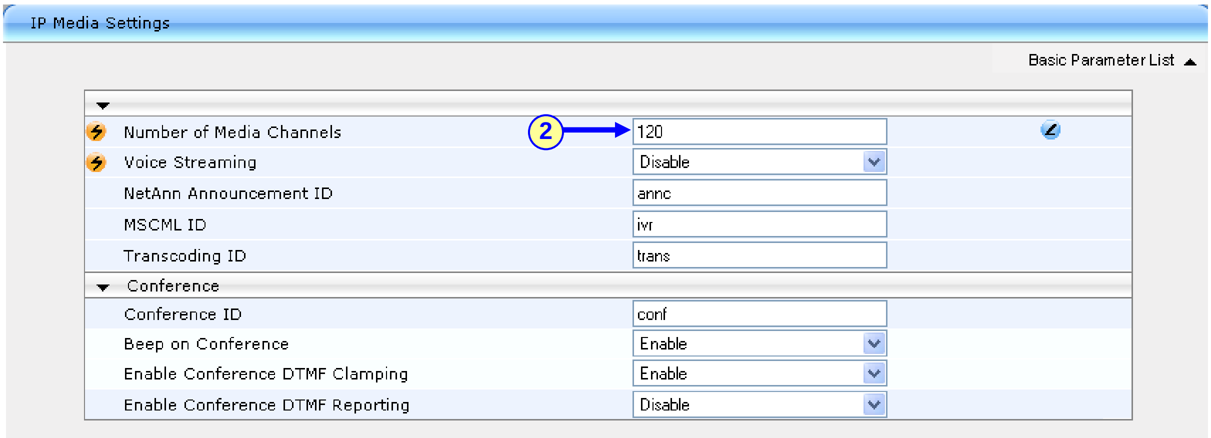
5.3 Step 3: Configure the Number of Media Channels

In order to reform the coder transcoding, you need to define DSP channels. The number of media channels represents the number of digital signaling processors (DSP) channels that the device allocates to IP-to-IP calls (the remaining DSP channels can be used for PSTN calls). Two IP media channels are used per IP-to-IP call. The maximum number of media channels available on the Mediant 1000 E-SBC device is 120 (i.e., up to 60 IP-to-IP calls). The maximum number of media channels available on the Mediant 3000 E-SBC Media Gateway device is 2016 (i.e., up to 1008 IP-to-IP calls).

➢ **To configure the number of media channels:**

1. Open the 'IP Media Settings' page (**Configuration** tab > **VoIP** menu > **IP Media** > **IP Media Settings**).

Figure 5-5: IP Media Channels Settings



2. In the 'Number of Media Channels', enter “120” to enable up to 60 IP-to-IP calls with transcoding. Click **Apply New Value**.

5.4 Step 4: Configure the Proxy Sets

This step describes how to configure the Proxy Sets. The Proxy Sets represent the IP addresses (or FQDN), which are required for communicating with the entities in the network:

- Proxy Set ID #1 is assigned with the IP address of AT&T IP Flexible Reach SIP Trunk.
- Proxy Set ID #2 is assigned with the IP address of Lync Mediation server.

These Proxy Sets are later assigned to IP Groups (see Section 5.5 on page 43).

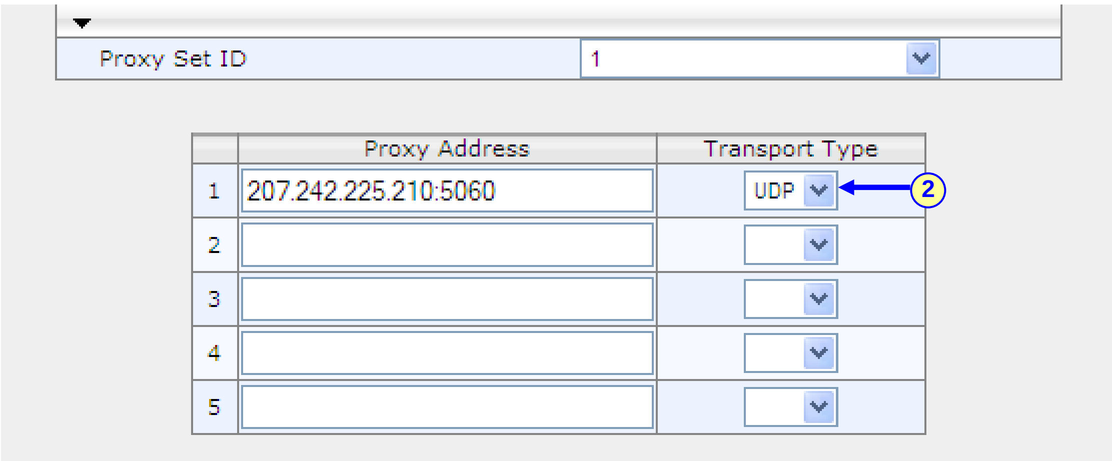
➢ **To configure proxy sets:**

1. Open the 'Proxy Sets Table' page (**Configuration** tab > **VoIP** menu > **Control Network** > **Proxy Sets Table**).
2. Configure the Proxy Set for AT&T IP Flexible Reach SIP Trunk:

   From the 'Proxy Set ID' drop-down list, select "1".

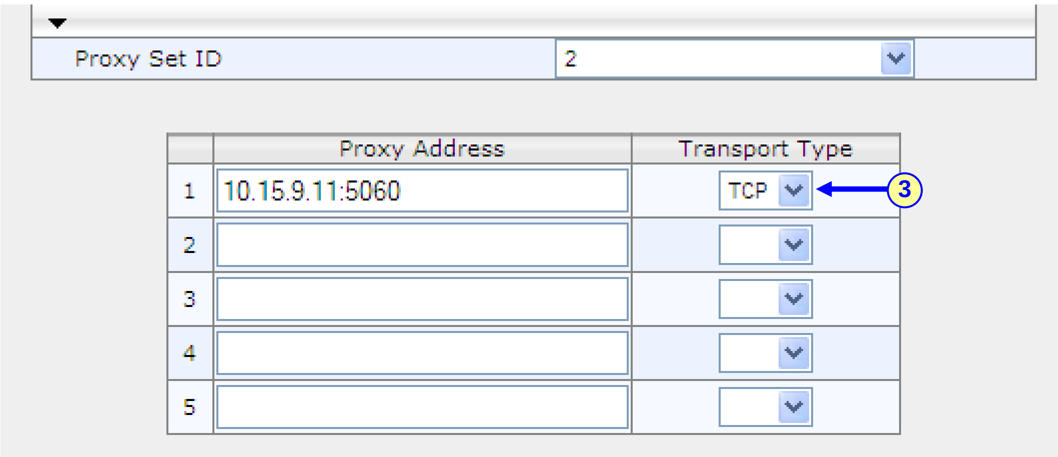
   a. In the 'Proxy Address' column, enter the IP address or FQDN of the AT&T IP Flexible Reach SIP Trunk and the listening port of the AT&T IP Flexible Reach SIP Trunk.
   b. From the 'Transport Type' drop-down list, corresponding to the IP address entered above, select "UDP".
   c. Repeat steps 'a' and 'b' as required for alternate AT&T IP Border Element (if used).

Figure 5-6: Proxy Set ID 1 for AT&T IP Flexible Reach SIP Trunk



3. Configure the Proxy Set for the Lync Mediation Server:
   a. From the 'Proxy Set ID' drop-down list, select "2".
   b. In the 'Proxy Address' column, enter the IP address or the FQDN and the listening port of the Lync Mediation Server.
   c. From the 'Transport Type' drop-down list corresponding to the IP address entered above, select "TCP" Transport Type.

Figure 5-7: Proxy Set ID 2 for Lync Mediation Server



5.5 Step 5: Configure the IP Groups

This step describes how to create IP groups. Each IP group represents a SIP entity in the device's network. You need to create IP groups for the following entities:

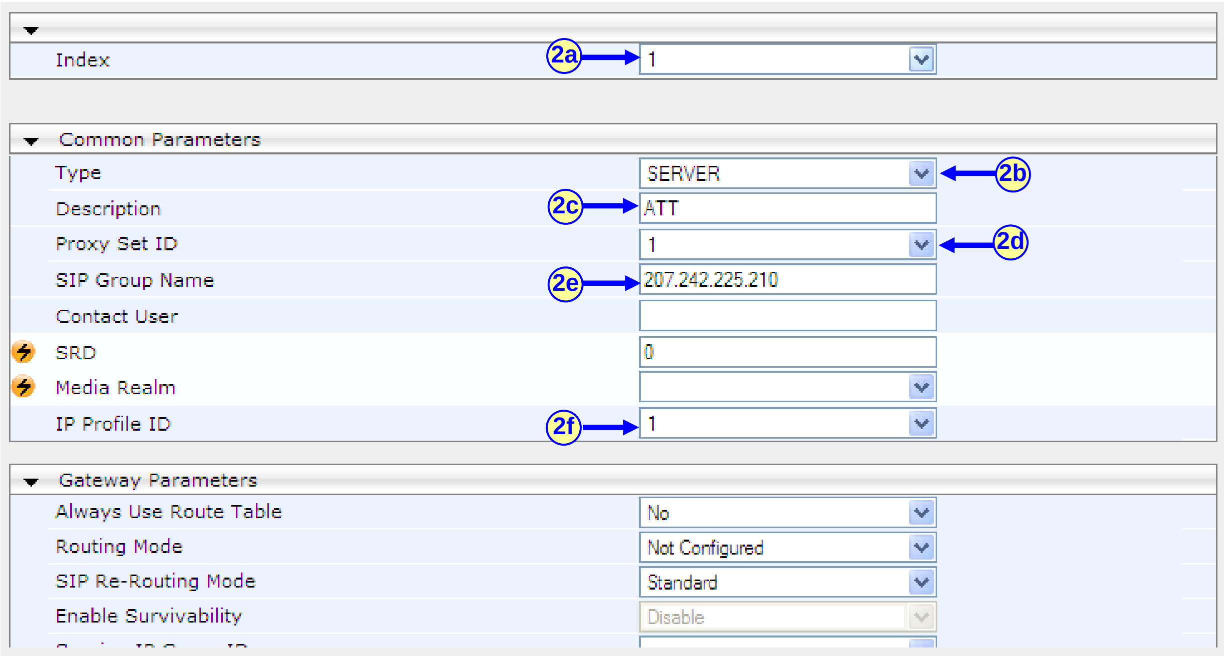
1. AT&T IP Flexible Reach SIP trunk
2. Lync Server 2010 – Mediation Server

These IP groups are later used by the IP2IP application for routing calls.

➢ To configure IP Groups:

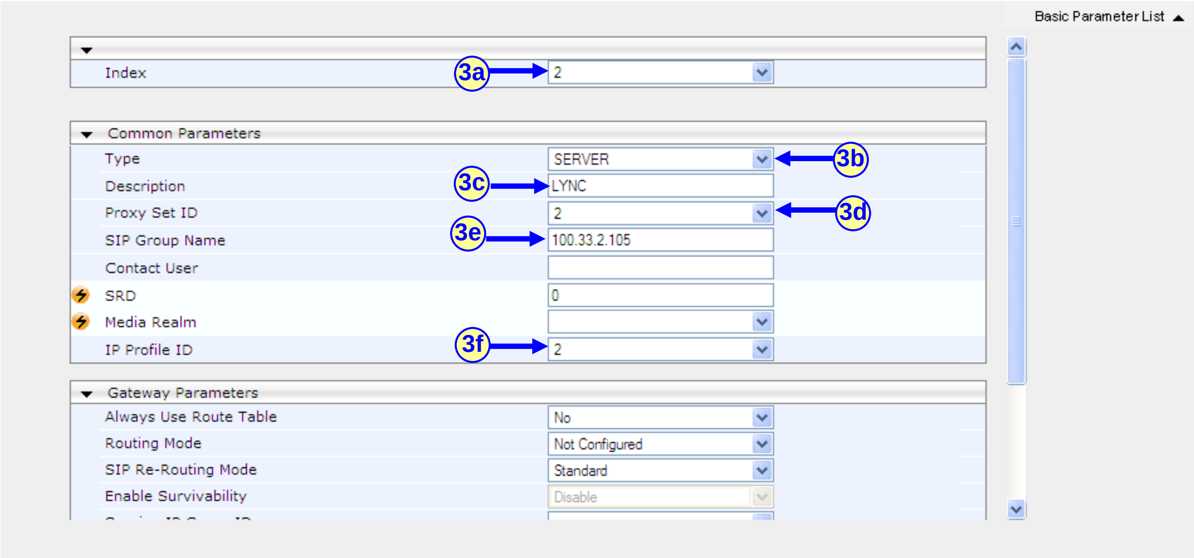
1. Open the 'IP Group Table' page (**Configuration** tab > **VoIP** menu > **Control Network**> **IP Group Table**).

Figure 5-8: IP Group 1 Table



2. Define IP Group #**1** for the AT&T IP Flexible Reach SIP Trunk as follows:
   a. IP Group Index '1'
   b. Type: "SERVER"
   c. Description: arbitrary name. (e.g., "ATT")
   d. Proxy Set ID: "1" (represents the IP address, configured in Section 5.4 on page 41, for communicating with this IP Group).
   e. SIP Group Name: The SIP Request-URI host name used in INVITE messages sent to the IP Group, or the host name in the From header of INVITE messages received from the IP Group. Enter the WAN IP address.
   f. IP Profile ID: "1": Different IP profile is used for the AT&T IP Flexible Reach SIP Trunk and the Mediation Server. See Section 5.7 on page 47.

Figure 5-9: IP Group 2 Table Page



3. Define IP Group #**2** for Mediation Server as follows:
   a. Select IP Group Index '2':
   b. **Type:** "SERVER"
   c. **Description:** <Free Description> (e.g., "Lync Mediation Server")
   d. Proxy Set ID: "2"
   e. **SIP Group Name:** The SIP Request-URI host name used in INVITE messages sent to the IP Group, or the host name in the From header of INVITE messages received from the IP Group. Enter the Gateway Name.
   f. **IP Profile ID:** "2" (see Section 5.7 on page 47).

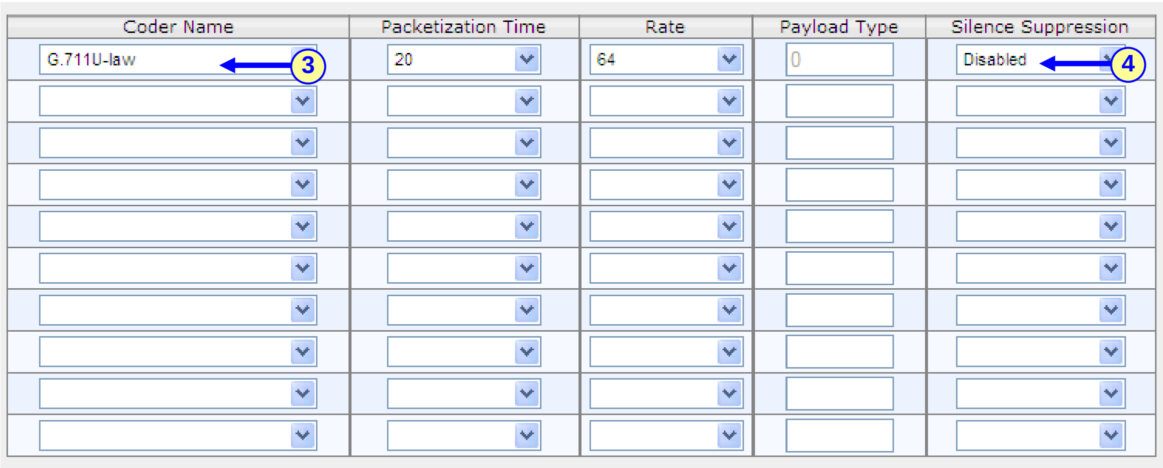
5.6 Step 6: Configure the Voice Coders

Since the Mediation Server supports both the G.711A-law and G.711U-law voice coders, while the AT&T Flexible Reach SIP trunk requires the G.711U-law coder, you can configure a single coder table reference for both services by utilizing the G711U-law coder. The Coder table is associated with an IP Profile. Both IP Profile indices 1 & 2 referenced in this document, reference 'Default Coder Table' (see Section 5.7 on page 47) which is associated with the IP Group (see Section 5.5 on page 43).

➢ **To configure Coder Table for Mediation server and AT&T Flexible Reach SIP Trunk:**

1. Open the 'Coders Table' page (**Configuration** tab > **VoIP** menu > **Coders And Profiles** > **Coders**).

Figure 5-10: Coder Group Table - Mediation Server



2. From the 'Coders Table' prepare to select via drop-down list, coder and attributes.
3. Select the G.711U-law coder, as shown in the figure above.
4. From 'Silence Suppression' drop-down list, select 'Enable' or 'Disabled' as shown in the figure above.

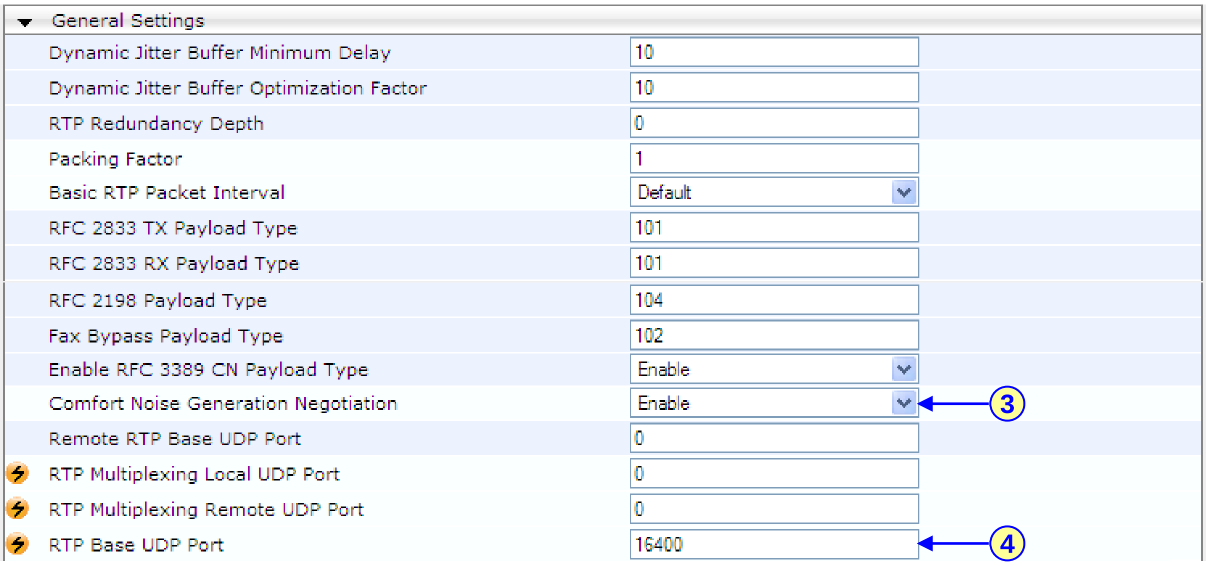
5.6.1 Step 7: Define Silence Suppression and Comfort Noise

Overall voice quality has been significantly improved for the Microsoft Lync 2010 environment. These improvements include suppression of typing noise during calls and improved generation of "comfort noise," which reduces hissing and smoothes over the discontinuous flow of audio packets. You may need to modify the Silence Suppression and Comfort Noise parameters to achieve this goal. Note that the Echo canceller is enabled by default.

➢ **To configure silence suppression parameters:**

1. Silence Suppression is configured per coder type. (See Section 5.6 on page 45 above to enable Silence Suppression per coder.)
2. Open the 'RTP/RTCP Settings' page (**Configuration** tab > **Media** menu > **RTP / RTCP Settings**).

Figure 5-11: RTP/RTCP Settings Page



3. From the 'Comfort Noise Generation Negotiation' drop-down list, select 'Enable'. This action enables negotiation and usage of Comfort Noise (CN).
4. Ensure 'RTP Base UDP Port' is '16400' to ensure that the RTP is within the port range of 16384-32767 (port range required by the Cisco Router with IP Flexible Reach-MIS/PNT/AVPN).
5. Click **Submit**.

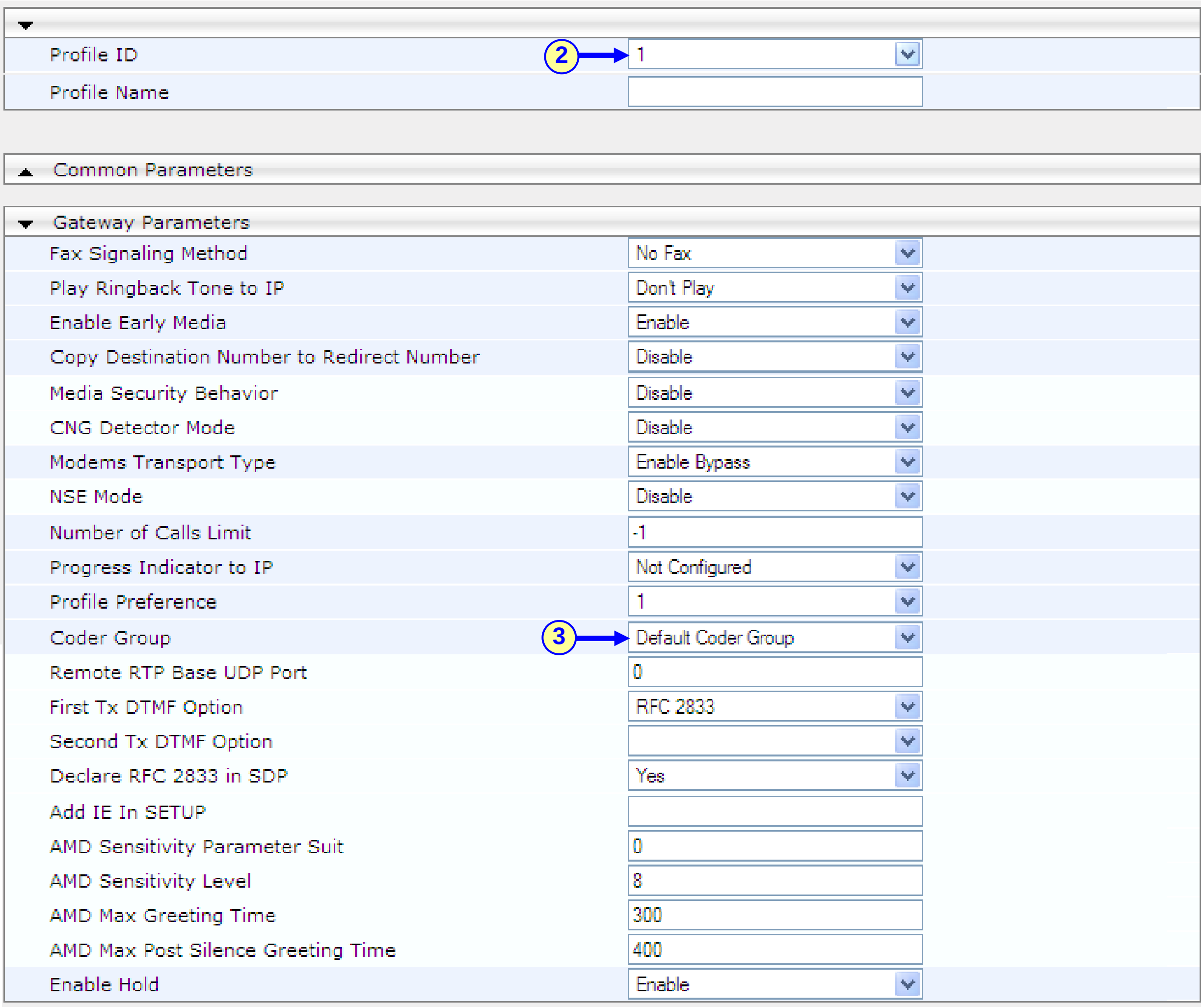
5.7 Step 8: Configure IP Profile Settings

This section describes how to configure the IP Profile Settings.

➢ To configure IP Profile for AT&T IP Flexible Reach:

1. Open the 'IP Profile Settings' page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **IP Profile Settings**).

Figure 5-12: IP Profile Page-AT&T IP Flexible Reach Server

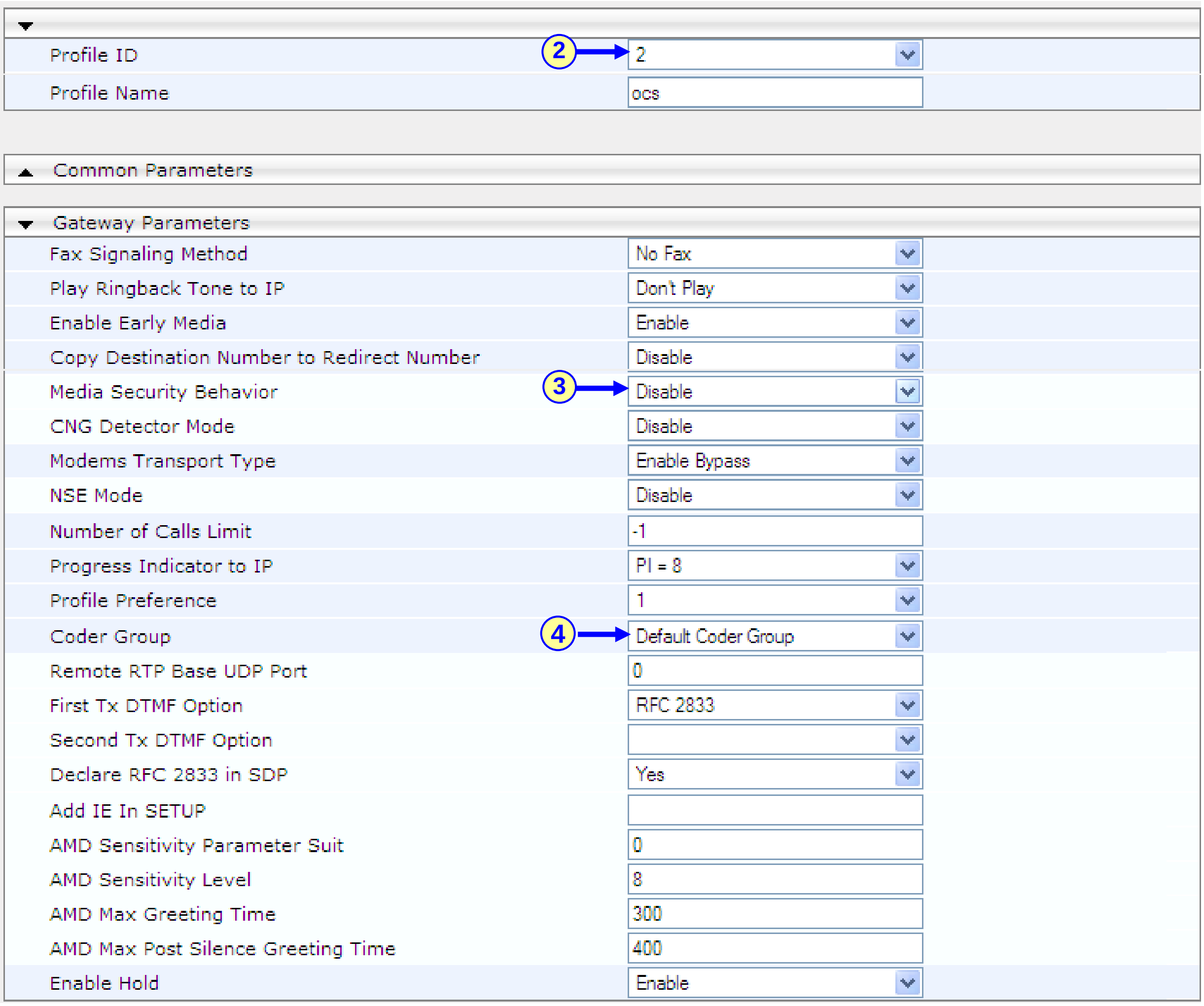


2. From the 'Profile ID' drop-down list, select '1'.
3. From the 'Coder Group' drop-down list, select 'Default Coder Group'.

➢ To configure IP Profile for Mediation Server:

1. Open the 'IP Profile Settings' page (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **IP Profile Settings**).

Figure 5-13: IP Profile Page-Mediation Server



2. From the 'Profile ID' drop-down list, select '2'.
3. From the 'Media Security Behavior' drop-down list, select one of the following options:
   - "Mandatory" if Mediation Server is configured to SRTP Required
   - "Preferable-Single media" if Mediation Server is configured to SRTP Optional.
   - "Disable" if the Mediation Server is configured to SRTP disabled.
4. From the 'Coder Group' drop-down list, select 'Default Coder Group'.

5.8 Step 9: Configure IP-to-IP Routing Setup

The E-SBC devices' IP-to-IP call routing capabilities is performed in two stages:

1. **Inbound IP Routing:** Recognizes the received call as an IP-to-IP call, based on the call's source IP address. This step is configured in the 'Inbound IP Routing Table'

2. **Outbound IP Routing:** Once recognized as an IP-to-IP call in the first stage (see above), the call is routed to the appropriate destination (i.e., IP address). This step is configured in the 'Outbound IP Routing Table'.

5.8.1 Configure Inbound IP Routing

This step defines how to configure the E-SBC device for routing inbound (i.e., received) IP-to-IP calls. The figure shown below illustrates three different call scenarios, corresponding to Index #1, Index #2 and Index#3 (described below).

➢ **To configure inbound IP routing:**

1. Open the 'Inbound IP Routing Table' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Routing** submenu > **IP to Trunk Group Routing**).

Figure 5-14: Inbound IP Routing Table Page

Routing Index	1-10
IP To Tel Routing Mode	Route calls before manipulation

	Dest. Host Prefix	Source Host Prefix	Dest. Phone Prefix	Source Phone Prefix	Source IP Address	->	Trunk Group ID	IP Profile ID	Source IPGroup ID
1	2		*	+1214291	10.15.9.11		-1	2	2
2	3	FE-Lync.Lync.local	*	*	10.15.9.11		-1	3	2
3	4		*	*	207.242.225.210		-1	1	1

2. **Index #1** configuration identifies all IP calls received from the Mediation Server as IP-to-IP calls and assigns them to the IP Group ID configured for the Lync Mediation Server as verified Lync assigned telephone numbers within a prefix range:
    - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
    - 'Source Phone Prefix': Enter the Lync assigned telephone prefix to screen for vaid direct IP-toIP calls.
    - 'Source IP Address': Enter the IP address of the Mediation server.
    - 'Trunk Group ID': Enter "-1" to indicate that these calls are IP-to-IP calls.
    - 'IP Profile ID: Enter '2' indicate the IP Profile for Mediation server.
    - 'Source IP Group ID': Enter "2" to assign these calls to the IP Group pertaining to the Mediation server.
3. **Index #2** configuration identifies all IP calls received from the Mediation Server in the event of a call forwarding Scenario (see section 5.10 on page 57) as IP-to-IP calls and assigns them to the IP Group ID configured for the Mediation server:
    - 'Source Host Prefix: Enter the Lync Front end FQDN – in case of call forwarding, the Source host in the incoming INVITE from the Mediation Server is the Lync Front End server FQDN, while for regular calls, the Source host is the Mediation Server FQDN.
    - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
    - 'Source IP Address': Enter the IP address of Mediation Server.
    - 'Trunk Group ID': Enter "-1" to indicate that these calls are IP-to-IP calls.
    - 'IP Profile ID: Enter '3' to indicate that the IP Profile supports the call forwarding scenario.
    - 'Source IP Group ID': Enter "2" to assign these calls to the IP Group pertaining to the Mediation server.
4. **Index #3** configuration identifies all IP calls received from AT&T IP Flexible Reach SIP Trunk as IP-to-IP calls and assigns them to the IP Group ID configured for the AT&T IP Flexible Reach SIP Trunk:
    - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
    - 'Source IP Address': Enter the IP address of AT&T IP Flexible Reach SIP Trunk.
    - 'Trunk Group ID': Enter "-1" to indicate that these calls are IP-to-IP calls.
    - 'IP Profile ID: Enter '1' indicate the IP Profile for AT&T IP Flexible Reach SIP Trunk.
    - 'Source IP Group ID': Enter "1" to assign these calls to the IP Group pertaining to AT&T IP Flexible Reach SIP Trunk.

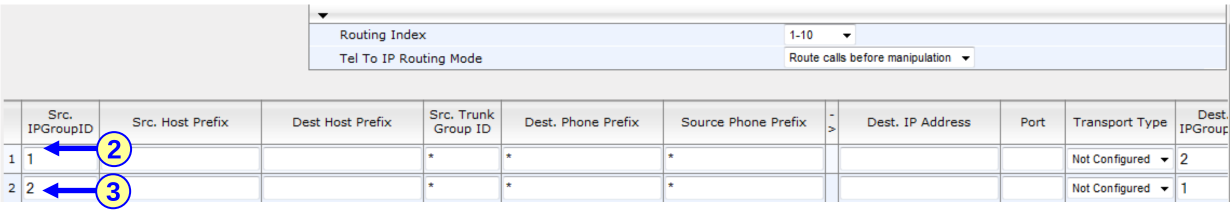
5.8.2 Configure Outbound IP Routing

This step defines how to configure the E-SBC device for outbound routing (i.e., sent) IP-to-IP calls. The figure shown below illustrates two different call scenarios, corresponding to Index #1 and Index #2 (described below).

➢ To configure outbound IP routing:

1. Open the 'Outbound IP Routing Table' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Routing** submenu > **Tel to IP Routing**).

Figure 5-15: Outbound IP Routing Table Page



2. **Index #1** defines routing of IP calls to the Lync 2010 Mediation server. All calls received from Source IP Group ID 1 (i.e., from the AT&T IP Flexible Reach SIP trunk) are routed to Destination IP Group ID 2 (i.e., to Lync 2010 Mediation server):
   - 'Source IP Group ID': Select "1" to indicate received (inbound) calls identified as belonging to the IP Group configured for the AT&T IP Flexible Reach SIP trunk.
   - 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
   - 'Source Phone Prefix': Enter the asterisk (*) symbol to indicate all callers.
   - 'Dest IP Group ID': Select "2" to indicate the destination IP Group to where the calls must be sent, i.e., to Lync 2010 Mediation server.
3. **Index #2** defines the routing of IP calls to the AT&T IP Flexible Reach SIP Trunk. All calls received from IP Group ID 2 (i.e., Lync 2010 Mediation server) are routed to Destination IP Group ID 1 (i.e., AT&T IP Flexible Reach SIP Trunk):
   - 'Source IP Group ID': Select "2" to indicate received (inbound) calls identified as belonging to the IP Group configured for the Lync 2010 Mediation Server 'Dest Phone Prefix': Enter the asterisk (*) symbol to indicate all destinations.
   - 'Source Phone Prefix': Enter the asterisk (*) symbol to indicate all callers.
   - 'Dest IP Group ID': Select "1" to indicate the destination IP Group to where the calls must be sent, i.e., to the AT&T IP Flexible Reach SIP Trunk.

5.9 Step 10: Configure Number Manipulation

The Manipulation Tables submenu allows you to configure number manipulation and mapping of NPI/TON to SIP messages. This submenu includes the following options:

- Dest Number IP->Tel. See Section 5.9.1 on page 53.
- Dest Number Tel->IP. See Section 5.9.1 on page 53.
- Source Number IP->Tel. See Section 5.9.2 on page 55.
- Source Number Tel->IP. See Section 5.9.2 on page 55.

5.9.1 Configure Destination Phone Number Manipulation

This section describes how to configure the destination phone number manipulation.

➢ **To configure Destination Phone Number Manipulation Table for IP -> Tel Calls Table:**

1. Open the 'Destination Phone Number Manipulation Table for IP -> Tel calls' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Manipulations** submenu > **Dest Number IP** >**Tel**).

Figure 5-16: Destination Phone Number Manipulation Table for IP -> Tel Calls Page

Index	Destination Prefix	Source Prefix	Source IP Address	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Num
1	+1	*	10.15.9.11	2	0	1		255
2	+0	*	10.15.9.11	1	0			255
3	+	*	10.15.9.11	1	0	1		255
4	1	*	10.15.9.11	1	0	1		255

- **Index #1** defines destination number manipulation of IP calls from Lync 2010 Mediation server. All calls received from IP address 10.15.9.11 (i.e., from Mediation Server) and the destination number prefix begins with '+1', Remove the '+1' from the Number, and add the prefix '1'.
- **Index #2** defines destination number manipulation of IP calls from Lync 2010 Mediation server. All calls received from IP address 10.15.9.11 (i.e., from Mediation Server) and the destination number prefix begins with '+0', Remove the '+' from the Number.
- **Index #3** defines destination number manipulation of IP calls from Lync 2010 Mediation server. All calls received from IP address 10.15.9.11 (i.e., from Mediation Server) and the destination number prefix begins with '+', Remove the '+' from the Number, and add the prefix '1'.
- **Index #4** defines destination number manipulation of IP calls from Lync 2010 Mediation server. All calls received from IP address 10.15.9.11 (i.e., from Mediation Server) and the destination number prefix begins with '1', Remove the '1' from the Number, and add the prefix '1'.

➢ To configure Destination Phone Number Manipulation Table for Tel -> IP Calls Table:

1. Open the 'Destination Phone Number Manipulation Table for Tel -> IP calls' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Manipulations** submenu > **Dest Number Tel->IP**).

Figure 5-17: Destination Phone Number Manipulation Table for Tel -> IP Calls Page

Index	Source Trunk Group	Source IP Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digit Leave
0	-1	1	+	*	0	0			255
1	-1	1	1	*	0	0	+		255
2	-1	1	3684891	*	7	0	+12142911002		255
3	-1	1	XXXXXXXXXX#	*	0	0	+1		255
4	-1	2	1511	*	1	0			255
5	-1	2	1911	*	1	0			255

- **Index #0** defines destination number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the destination number prefix begins with '+', do not perform any changes to the number.
- **Index #1** defines destination number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the destination number prefix begins with '1', add the '+' prefix to the number.
- **Index #2** defines destination number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the destination number prefix begins with '3684891' as a 7 digit private number, remove the 7 digits, and add the '+12142911002' as the new 10 digit number.
- **Index #3** defines destination number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the destination number length is 10 digit number, add the '+1' prefix to the number.
- **Index #4** defines destination number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 2 (i.e., from Lync 2010 Mediation Server) and the destination number is '1511', remove the '1' prefix from the left.
- **Index #5** defines destination number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 2 (i.e., from Lync 2010 Mediation Server) and the destination number is '1911', remove the '1' prefix prefix from the left.

5.9.2 Configure Source Phone Number Manipulation

➢ **To configure Source Phone Number Manipulation Table for IP -> Tel Calls Table:**

1. Open the ‘Source Phone Number Manipulation Table for IP -> Tel calls’ page (**Configuration** tab > **VoIP menu** > **GW and IP to IP** > **Manipulations** submenu > **Source Number IP** >**Tel**).

Figure 5-18: Source Phone Number Manipulation Table for IP -> Tel Calls Page

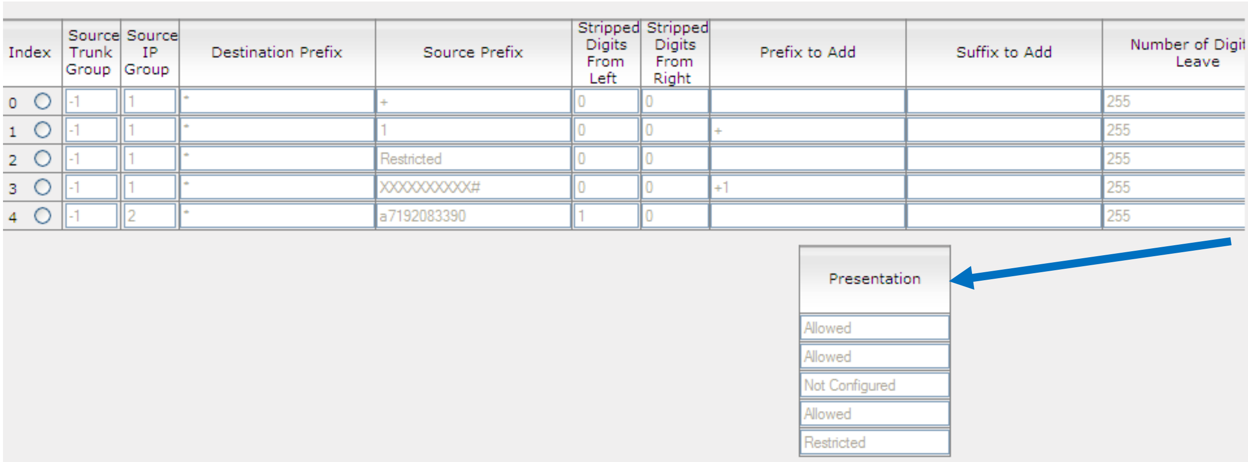
Index	Destination Prefix	Source Prefix	Source IP Address	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digits to Leave	NPI	TON	Presentation
1	*	+1	10.15.9.11	2	0			255	Not Configured	Not Configured	Allowed
2	*	+	10.15.9.11	1	0			255	Not Configured	Not Configured	Allowed
3	*	1	10.15.9.11	1	0			255	Not Configured	Not Configured	Allowed
4	*	anonymous	10.15.9.11	20	0	a7192083390		255	Not Configured	Not Configured	Not Configured

- **Index #1** defines Source number manipulation of IP calls from Lync Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Lync Mediation Server) and the source number prefix begins with '+1', remove the ‘+1’ from the number.
- **Index #2** defines Source number manipulation of IP calls from Lync Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Lync Mediation Server) and the source number prefix begins with '+', remove the ‘+’ from the number.
- **Index #3** defines Source number manipulation of IP calls from Lync Mediation Server. All calls received from IP address 10.15.9.11 (i.e., from Lync Mediation Server) and the source number prefix begins with '1', remove the ‘1’ from the number.
- **Index #4** defines Source number manipulation of anonymous calls from Lync Mediation Server. Anonymous calls received from IP address 10.15.9.11 (i.e., from Lync Mediation Server) replace the ‘anonymous’ caller ID with a modified well known number i.e. a7192083390. This manipulation is performed to create a well known number in the P-Asserted-Identity header. Without this number, the AT&T IP Flexible Reach SIP Trunk rejects the call. See later on for the Source Number manipulation Tel->IP manipulation rule that restricts the caller ID for an anonymous call on page 29 within Source Phone Number Manipulation Table for Tel -> IP index #4.

➢ To configure Source Phone Number Manipulation Table for Tel -> IP Calls Table:

1. Open the 'Source Phone Number Manipulation Table for Tel -> IP calls' page (**Configuration** tab > **VoIP menu** > **GW and IP to IP** > **Manipulations** submenu > **Source Number Tel** > **IP**).

Figure 5-19: Source Phone Number Manipulation Table for Tel -> IP Calls Page



Index	Source Trunk Group	Source IP Group	Destination Prefix	Source Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digit Leave	Presentation
0	-1	1	*	+	0	0			255	Allowed
1	-1	1	*	1	0	0	+		255	Allowed
2	-1	1	*	Restricted	0	0			255	Not Configured
3	-1	1	*	XXXXXXXXXX#	0	0	+1		255	Allowed
4	-1	2	*	a7192083390	1	0			255	Restricted

- **Index #0** defines Source number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the Source number prefix begins with '+', do not perform any changes to the number.
- **Index #1** defines Source number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the Source number prefix begins with '1', Add a '+' as a prefix to the number.
- **Index #2** defines Source number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the Source number prefix is set as 'Restricted', do not perform any changes to the call.
- **Index #3** defines Source number manipulation of IP calls from AT&T IP Flexible Reach SIP Trunk. All calls received from Source IP Group 1 (i.e., from AT&T IP Flexible Reach SIP Trunk) and the Source number length is 10 digit number, add the '+1' prefix to the number.
- **Index #4** defines Source number manipulation of anonymous calls from Microsoft Lync environment. All calls received from Source IP Group 2 (and the Source number was modified to a7192083390 (which is a specially modified well known number that was inserted for the anonymous caller ID on the source number manipulation IP->Tel above), the 'a' is removed and the presentation should be set to 'restricted'. To simply mark all calls as Private calls, use an asterisk '*' in the Source Prefix field. Individual Telephone Numbers or ranges can also be set in this manner as well.

5.10 Step 11: Configure IP Profile for Call Forwarding

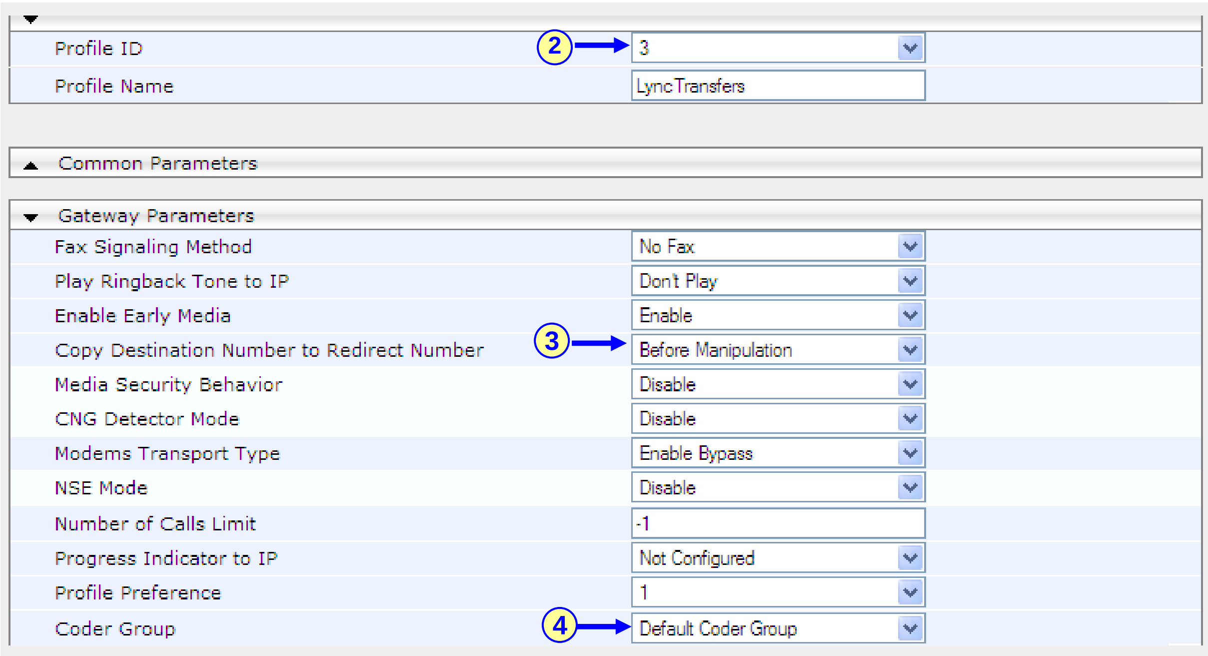
One of the challenges with the integration of the Microsoft Lync 2010 server and the Flexible Reach SIP Trunk is the implementation of call forwarding. Since the Microsoft Lync client forwards the call back to the SIP Trunk, it does not provide any information in the forwarded INVITE (such as Diversion header) informing that this call has been forwarded. Consequently, it is necessary to configure a special IP Profile that adds the diversion header toward the SIP trunk in the event of a call forwarding scenario.

This profile is later associated to the routing table in the event of a call forwarding scenario (see section 5.8.1 on page 49).

➢ To configure IP Profile for call forwarding:

1. Open the 'IP Profile Settings' page (**Configuration tab** > **VoIP menu** > **Coders and Profiles** > **IP Profile Settings**).

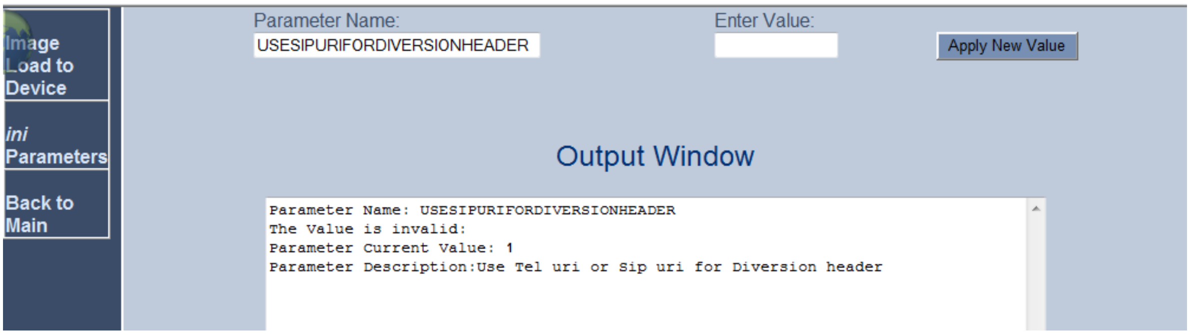
Figure 5-20: IP Profile Settings for Call Forwarding "numbers"



2. From the 'Profile ID' drop-down list, select '3'.
3. From the 'Copy Destination Number to Redirect Number' drop-down list, select 'Before Manipulation'; this parameter adds the Diversion Header to the INVITE in event of a call forwarding scenario.
4. From the 'Coder Group' drop-down list, select 'Default Coder Group'.

5. Open the 'Admin" page, by appending the case-sensitive suffix 'AdminPage' to the Media Gateway's IP address in your Web browser's URL field (e.g., http://10.15.4.15/AdminPage).
6. On the left pane, click ***ini*** Parameters.

Figure 5-21: Output Window



7. In the 'Parameter Name' field, enter the parameter **USESIPURIFORDIVERSIONHEADER**. In the 'Enter Value' field, enter "1".
8. Click **Apply New Value**.

5.10.1 Configure Redirect Number Manipulation

In the event of a call forwarding scenario, a Diversion header needs to be added to the INVITE towards the Flexible Reach SIP Trunk (as configured in Section 5.10 above). In this case, the E-SBC copies the Destination number to the Redirect number and adds this number to the Diversion header. In order to have a well known number in the Diversion header (for Flexible Reach SIP Trunk), a manipulation rule should be defined to replace the redirect number to a well known number.

➢ **To configure redirect number Tel -> IP Table:**

1. Open the 'Redirect Number Tel -> IP' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Manipulations** submenu > **Redirect Number Tel** > **IP**).

Figure 5-22: Redirect Number Tel -> IP Page

Index	Source Trunk Group	Source IP Group	Destination Prefix	Redirect Prefix	Stripped Digits From Left	Stripped Digits From Right	Prefix to Add	Suffix to Add	Number of Digit Leave
1	-1	-1	*	*	20	0	7323680193		255

- Index **#1** defines redirect number manipulation for the call forwarding scenario.

The redirect number is changed to a well known number i.e. 7323680193.

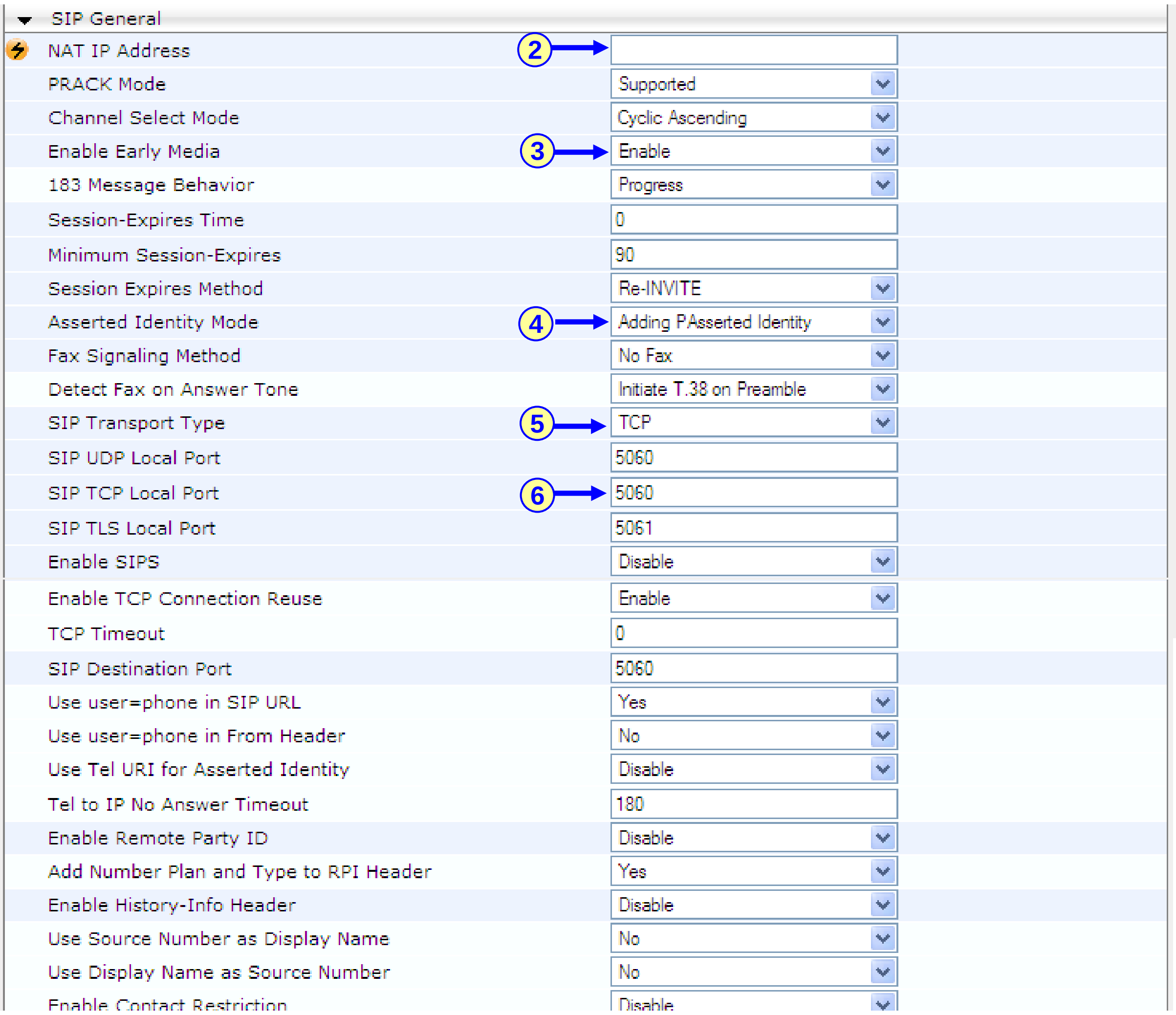
5.11 Step 12: Configuring SIP General Parameters

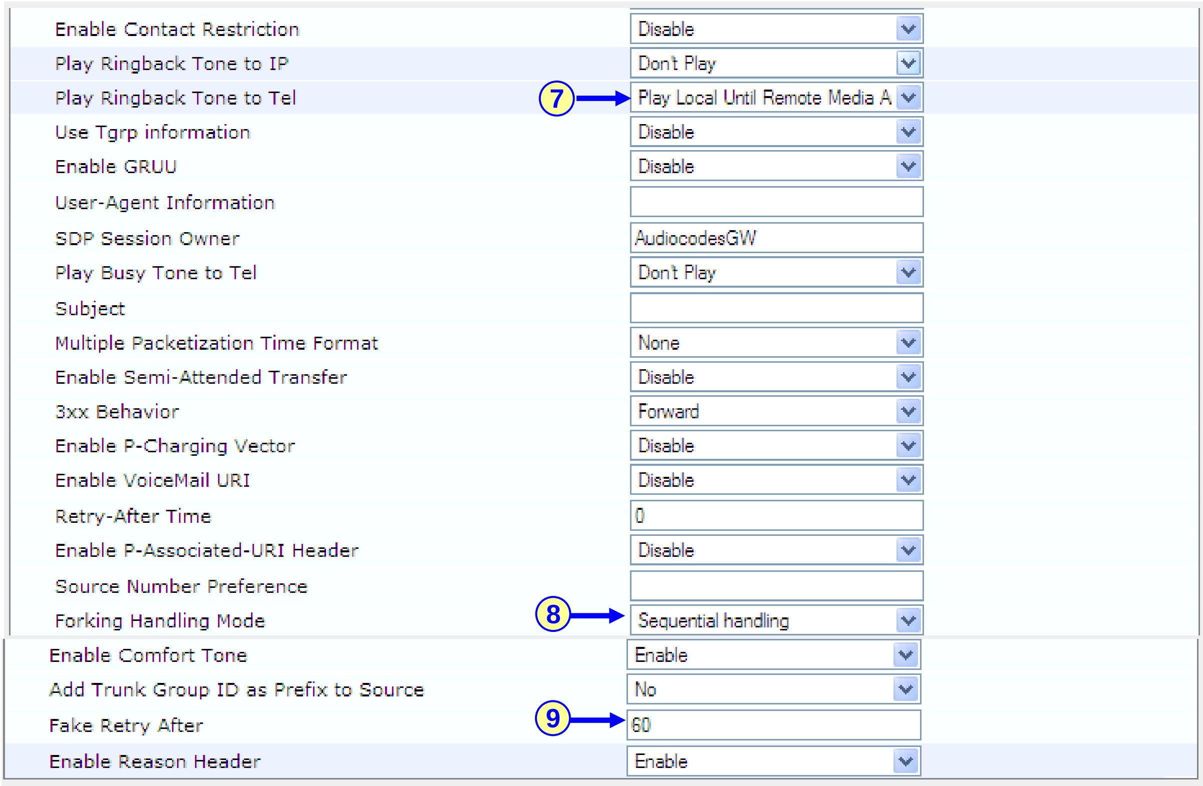
This section describes how to configure the SIP general parameters.

➢ To configure general SIP parameters:

1. Open the 'SIP General Parameters' page (**Configuration** tab > **VoIP** menu > **SIP Definitions** submenu > **General Parameters**).

Figure 5-23: SIP General Parameters Page





2. In the 'NAT IP Address' field, ensure this field is empty. It is not used for the Single LAN Interface solution. This field is only used when the WAN Interface is utilized with a Global (public) IP address of the E-SBC device to enable the static NAT between the E-SBC device and the Internet.
3. From the 'Enable Early Media' drop-down list, select 'Enable' to enable early media.
4. From the 'Asserted Identity Mode' drop-down list, select 'Adding PAsserted Identity'.
5. From the 'SIP Transport Type' drop-down list, select 'TCP' in case the Mediation Server is configured to use TCP transport Type.
6. In the 'SIP TCP Local Port' field, enter '5060'; this port is the listening E-SBC device port for TCP transport type. This port must match the transmitting port of the Mediation Server.
7. From 'Play Ringback Tone to Tel' drop-down list, select 'Play Local Until Remote Media Arrive'. Plays the RBT according to the received media. If a SIP 180 response is received and the voice channel is already open (due to a previous 183 early media response or due to an SDP in the current 180 response), the E-SBC device plays a local RBT if there are no prior received RTP packets. The E-SBC device stops playing the local RBT as soon as it starts receiving RTP packets. At this stage, if the E-SBC device receives additional 18x responses, it does not resume playing the local RBT.
8. From the 'Forking Handling Mode' drop-down list, select 'Sequential handling'; this parameter determines whether18x with SDP is received. In this case, the E-SBC device opens a voice stream according to the received SDP. The E-SBC device re-opens the stream according to subsequently received 18x responses with SDP.

9. In the 'Fake Retry After' field, enter '60' sec. This parameter determines whether the E-SBC device, upon receipt of a SIP 503 response without a Retry-After header, behaves as if the 503 response included a Retry-After header and with the period (in seconds) specified by this parameter.

5.12 Step 13: Defining Reasons for Alternative Routing

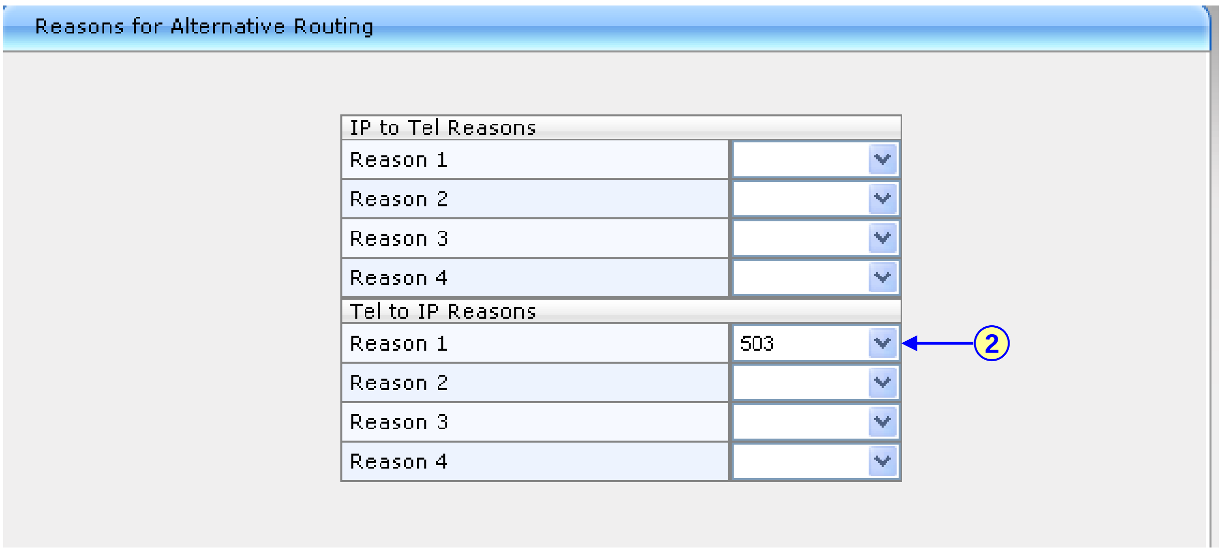
A 503 SIP response from the Mediation Server to an INVITE must cause the E-SBC device to perform a failover. In other words, if the Lync Mediation Server primary proxy server is not responding, an attempt is made to establish communication with the secondary proxy server. For this event to occur, you need to perform the following actions:

- Configure the Reasons for Alternative Routing for Tel-to-IP calls to '503 SIP response'.
- Configure the Lync Mediation Proxy Set for redundancy purposes.

➢ To define SIP Reason for Alternative Routing:

1. Open the 'Reasons for Alternative Routing' page (**Configuration** tab > **VoIP** menu > **GW and IP to IP** > **Routing** submenu > **Alternative Routing Reasons**).

Figure 5-24: Reasons for Alternative Routing Page



2. Under the Tel to IP Reasons group, for Reason 1, select '503'.
3. Click **Submit**.
4. Open the 'Proxy & Registration' page (**Configuration** > **VoIP** > **SIP Definitions** > **Proxy & Registration**) and configure the 'Redundant Routing Mode' parameter to 'Proxy' as shown below in Figure 5-25. This will allow entry back into the Proxy Set table for the next available route. This redundant route is configured in the next step (on Proxy Set ID 2, see Figure 5-26 below).

5. Open the 'Proxy Sets Table' page (**Configuration** tab > **VoIP** menu > **Control Network** > **Proxy Sets Table**). Configure the Proxy Set for the Lync Mediation Server:

   From the 'Proxy Set ID' drop-down list, select "2".

   a. In the 'Proxy Address' column, enter a second IP address or the FQDN and the listening port of the secondary Lync Mediation Server.

   b. From the 'Is Proxy Hot Swap' drop-down list, select "Yes".

6. Click **Submit**.

Figure 5-25: 'Proxy & Registration' Page

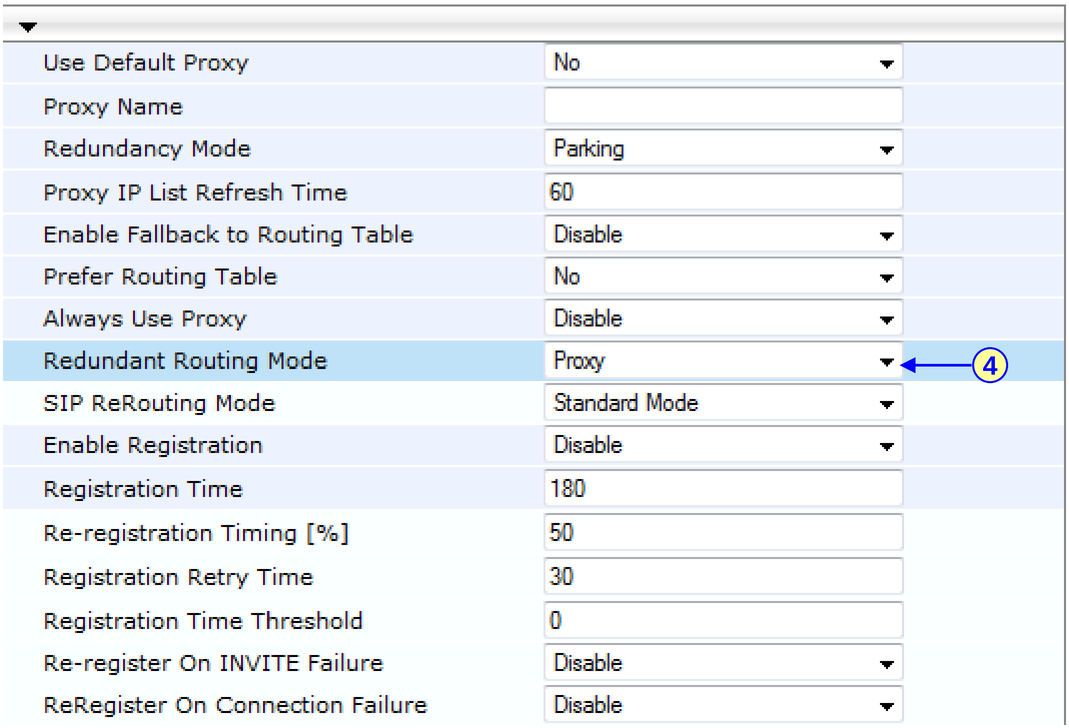
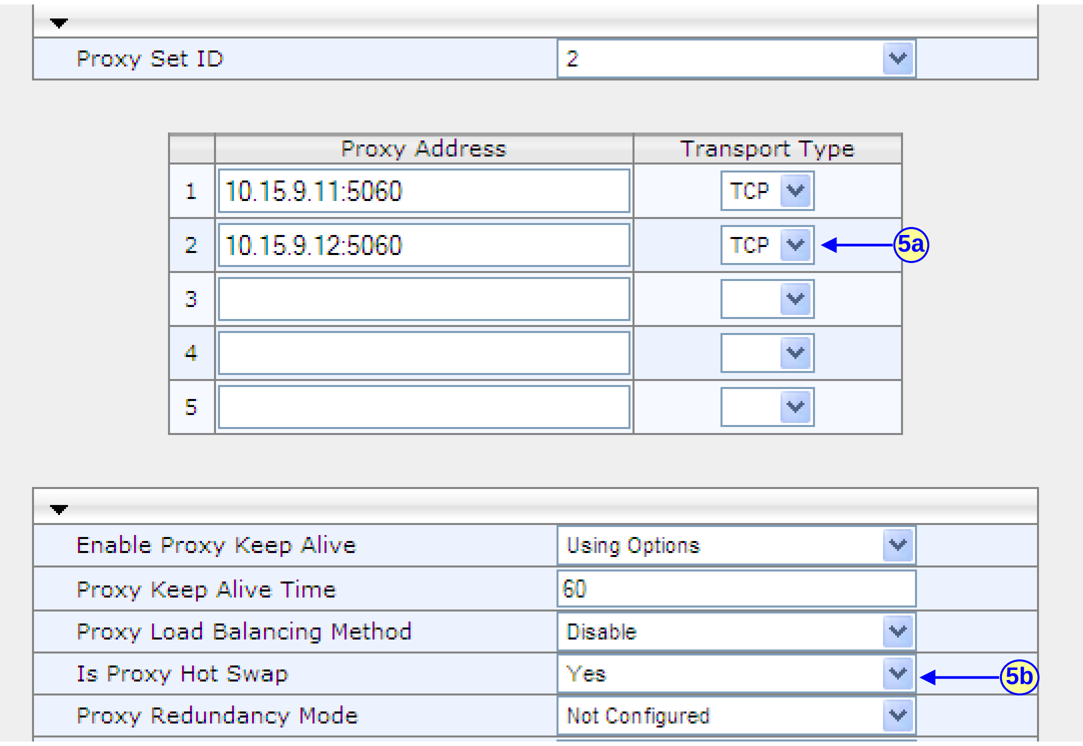


Figure 5-26: Proxy Set ID 2 for Lync Mediation Server



6 Troubleshooting

This section should provide some tips for troubleshooting problems, including troubleshooting commands and contact numbers within Vendor X's Company for trouble escalation.

6.1 Debugging Procedures

This section discusses the following debugging procedures:

- Case Reporting Procedures. See section 6.1.1 below.
- Syslog. See section 6.1.2 on page 66.
- Wireshark Network Sniffer. See section 6.1.3 on page 68.

6.1.1 Case Reporting Procedures

When reporting a problem to AudioCodes' Technical Support department, the following information should be provided:

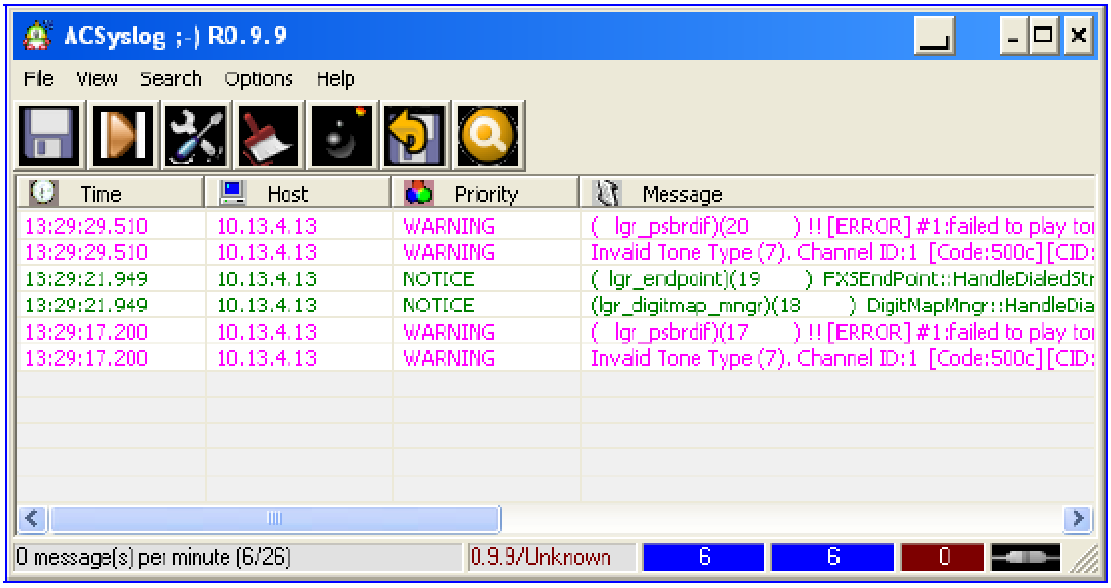
- Basic information (required for all types of problems):
  - Problem description (nature of failure, symptoms, call direction, etc.)
  - Network diagram
  - *ini* configuration file (downloaded to your PC from the device using the Web interface)
  - Syslog trace (without missing messages)
  - Unfiltered IP network trace using the Wireshark application

  (Note: If you are unable to collect all the network traffic, then at least collect the mandatory protocols SIP, RTP, and T38.)
- Advanced information (if required upon request):
  - PSTN message traces - for PSTN problems
  - Media stream traces - for problems related to voice quality, modem\fax, DTMF detection, etc.

6.1.2 Syslog

Syslog is a standard for forwarding log messages in an IP network. A syslog client, embedded in the device sends error reports/events generated by the device to a remote Syslog server using IP/UDP protocol. This information is a collection of error, warning and system messages that record every internal operation of the device. You can use the supplied AudioCodes proprietary Syslog server "ACSyslog" (shown in the figure below) or any other third-party Syslog server for receiving Syslog messages.

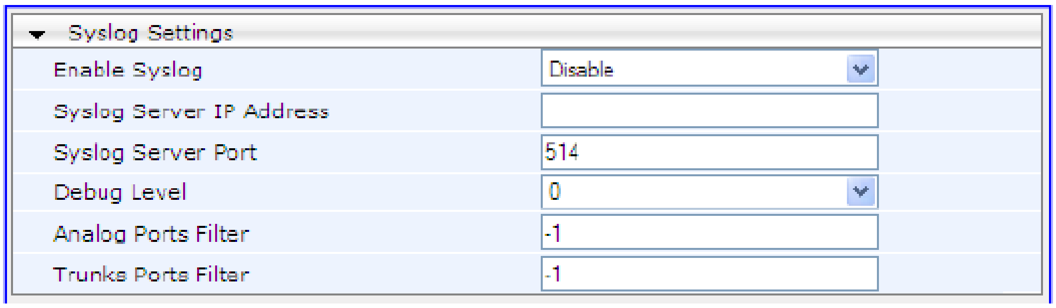
Figure 6-1: AudioCodes' Proprietary Syslog Server



➢ To activate the Syslog client on the device using the Web interface:

1. Open the 'Syslog Settings' page (**Configuration** tab > **System** menu > **Syslog Settings**).
2. In the 'Syslog Server IP Address' field, enter the IP address of the Syslog server (*ini* file parameter SyslogServerIP).
3. From the 'Enable Syslog' drop-down list, select 'Enable' to enable the device to send syslog messages to a Syslog server (defined in Step 2).

Figure 6-2: Enabling Syslog



4. From the 'Debug Level' drop-down list, select '5' if debug traces are required.

   To enable syslog reporting, using the *ini* file, load an *ini* file to the device with the following settings:

```
[Syslog]
SyslogServerIP = 192.168.2.35
EnableSyslog = 1
SyslogServerPort = 514
GWDebugLevel = 5
```

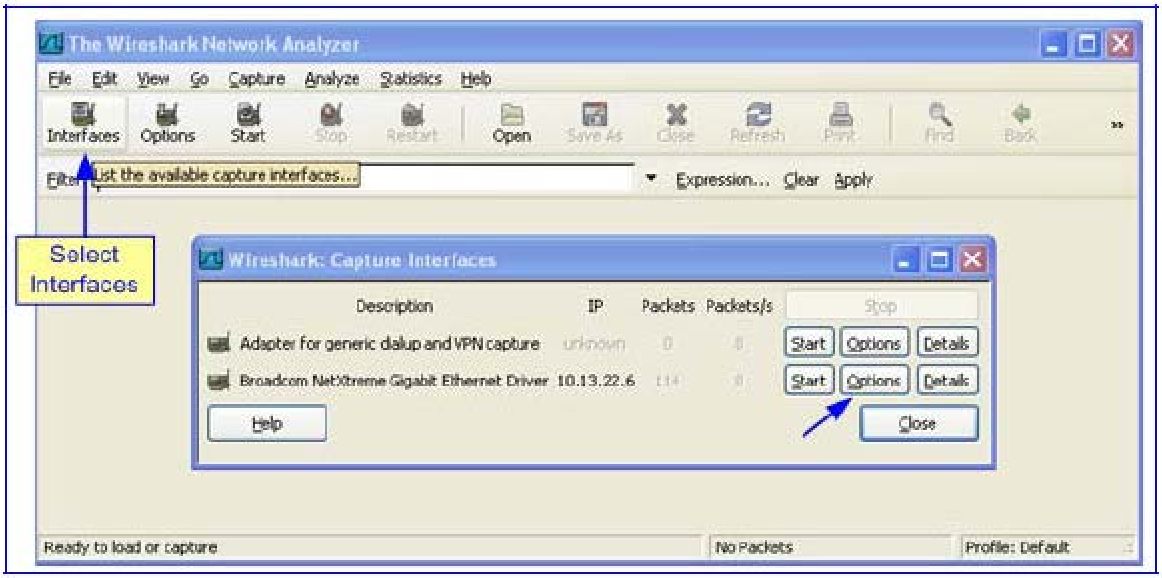
6.1.3 Wireshark Network Sniffer

Wireshark is a freeware packet sniffer application that allows you to view the traffic that is being passed over the network. Wireshark can be used to analyze any network packets. Wireshark can also be used to analyze RTP data streams and extract the audio from the data packets (only for G.711). The audio can be saved as a *.pcm file.

➢ **To record traffic that is sent to / from the device:**

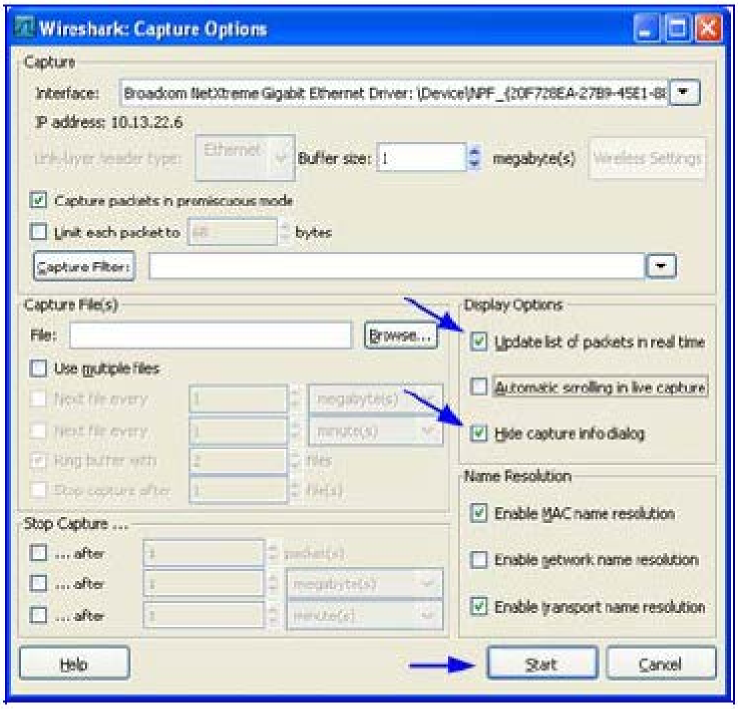
1. Install Wireshark on your PC. (You can download it from the following Web site: http://www.wireshark.org/
2. Connect the PC and the device to the same hub.
3. If you are using a switch, use a switch with port mirroring for the port to which the Wireshark is connected.
4. Start Wireshark.
5. Select the network interface that is currently being used by the PC - on the toolbar,click **Interfaces**, and then in the 'Capture Interfaces' dialog box, click the **Options** button corresponding to the network interface:

Figure 6-3: Selecting Interface Currently used by the PC



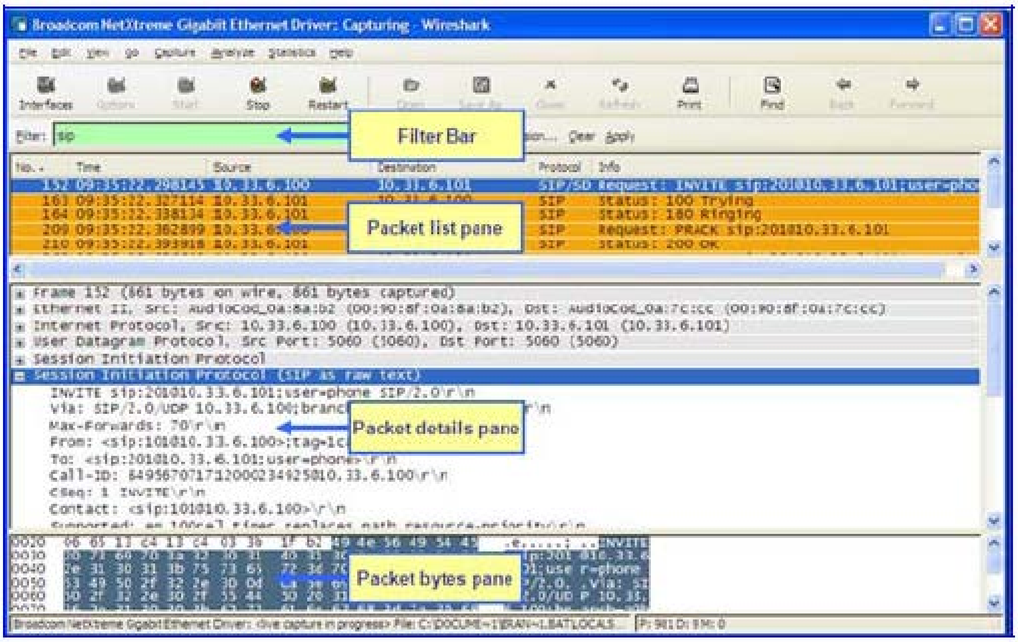
6. In the 'Capture Options' dialog box, select the desired display options:

Figure 6-4: Configuring Wireshark Display Options



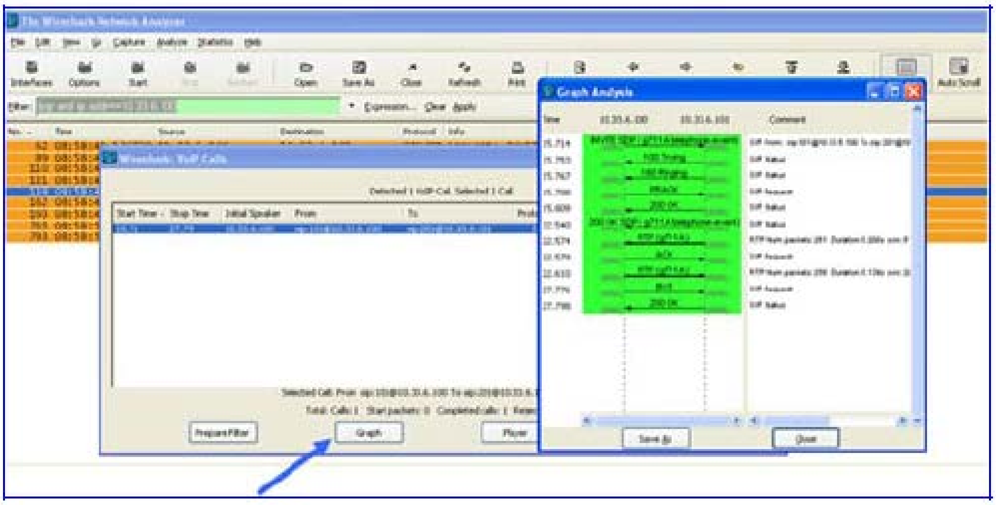
7. Click **Start**.

Figure 6-5: Captures Packets



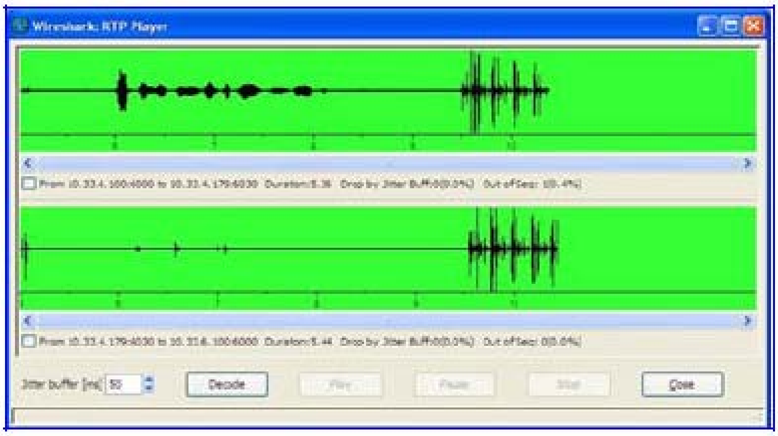
8. To view VoIP call flows, from the **Statistics** menu, choose **VoIP Calls**. You can view the statistics in graph format by clicking **Graph**.

Figure 6-6: Viewing VoIP Call Flows



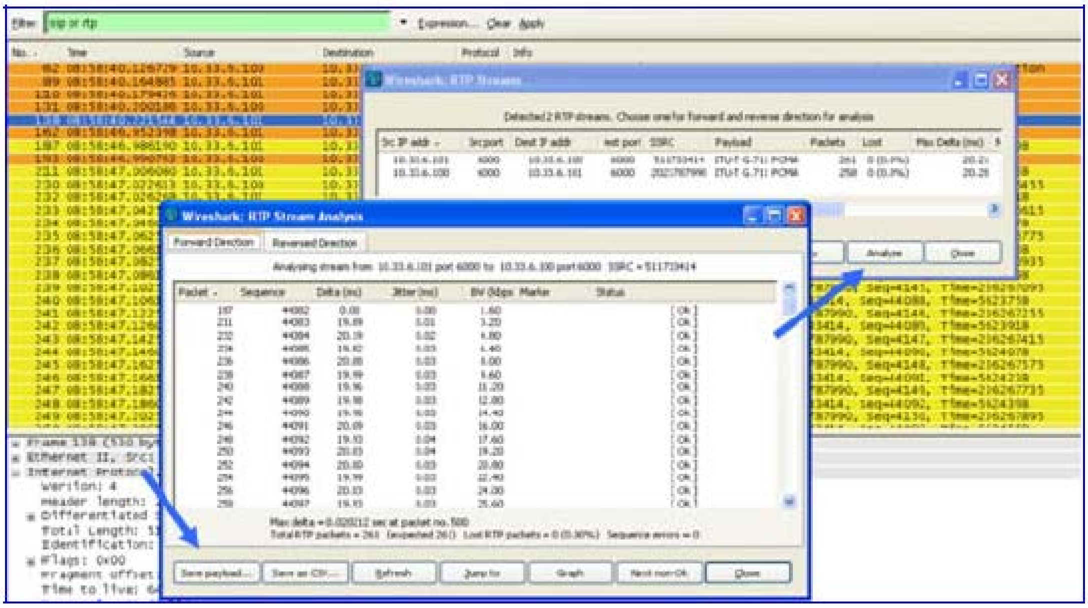
9. To play G.711 RTP streams, click the **Player** button.

Figure 6-7: Playing G.711 RTP Streams



10. To analyze the RTP data stream and extract the audio (which can be played using programs such as CoolEdit) from the data packets (only for G.711), from the **Statistics** menu, point to **RTP**, and then choose **Stream Analysis**.

Figure 6-8: Analyzing the RTP Data



a. Save the audio payload of the RTP stream to a file.

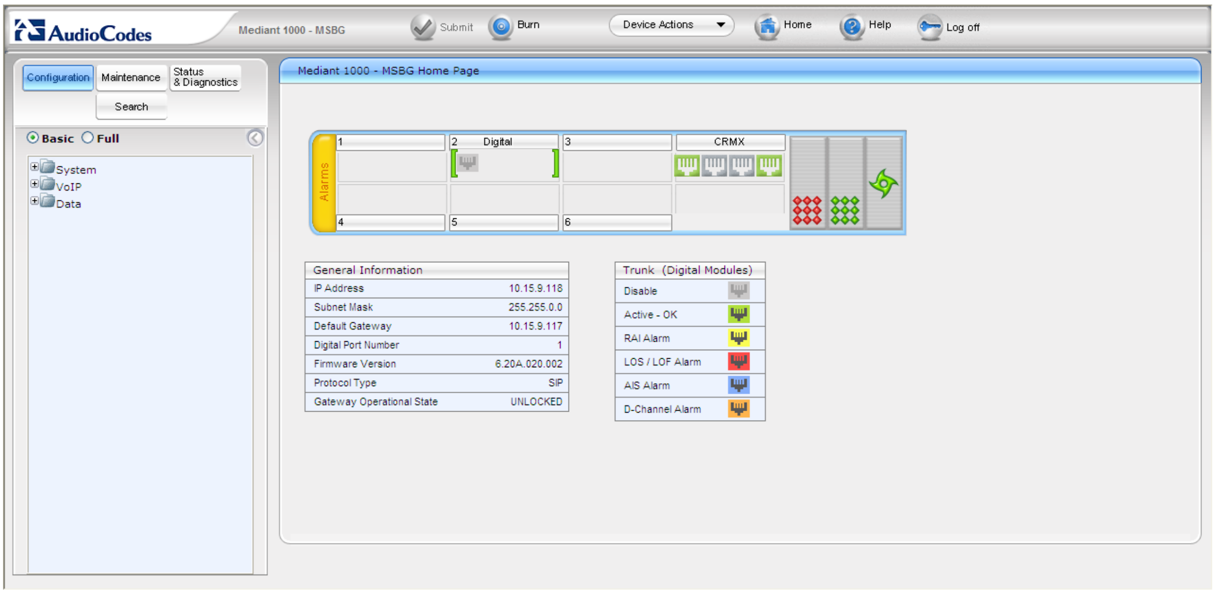
b. Save the Payload as a *.pcm file.

c. Select the ‘forward’ option.

6.2 Verifying Firmware

To verify the firmware load actively running on the device, log into the device and view the firmware version on the product homepage as shown in the figure below.

Figure 6-9: Viewing active firmware version



Reader's Notes

Configuration Note

Connecting AT&T's IP Flexible Reach - MIS/PNT/AVPN SIP Trunking Service to Microsoft® Lync Server 2010 Using AudioCodes' Mediant 800 E-SBC, Mediant 1000 MSBG E-SBC and Mediant 3000 E-SBC Media Gateway

Document #: LTRT-38100

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